Exhibit A

IN THE UNITED STATES DISTRICT COURT
FOR THE NORTHERN DISTRICT OF CALIFORNIA

EPL HOLDINGS, LLC Plaintiff-Counterclaim Defendant, v. APPLE INC. Defendant-Counterclaimant.	Case No. 3:12-cv-04306 (JST)
And Related Counterclaims	

DECLARATION OF PROF. JULIUS O. SMITH, PH.D. REGARDING CLAIM CONSTRUCTION FOR U.S. PATENT NOS. 5,175,769; 7,683,903; 8,345,050; AND 8,384,720

I. INTRODUCTION

1. I have been retained as an expert in this case by Apple Inc. ("Apple") to provide my opinions regarding U.S. Patent No. 5,175,769 to Hejna et al. ("the '769 Patent") and U.S. Patent Nos. 7,683,903; 8,345,050; and 8,384,720 to Goldhor et al. ("the '903 Patent," "the '050 Patent," and "the '720 Patent" respectively; collectively, "the '903 Patent Family"). It is my opinion that a person of ordinary skill in the art would have understood the claim terms of the '769 Patent and the '903 Patent Family to have the meanings as discussed below at the time those patent applications were filed.

2. I understand that EPL Holdings, LLC ("EPL") alleges that Apple has infringed claims 1, 2, 10, 11, and 19 of the '769 patent and that Apple infringes claims 1, 4, 6, 12, 13, 16, and 22 of the '903 patent; claims 1, 4, 5, 8, 10, 13, 16, 17, 20, 22, 25, 26, 27, 28, 29, 30, 31, 32, 33, 36, 37, 38, 39, 40, 41, 42, 43, 44, 47, 48, 49, and 50 of the '050 Patent; and claims 1, 6, 7, 10, and 11 of the '720 Patent (collectively, "the Asserted Claims"). I have not formed any opinions about whether Apple's accused products practice the limitations of the asserted claims, but reserve the right to do so at a later date.

3. I understand the parties have exchanged proposed constructions of certain terms in the Asserted Claims along with extrinsic support for those constructions. I have reviewed the '769 Patent and the '903 Patent Family, each patent's corresponding file history, and the parties' proposed constructions as provided in their Exchange Of Preliminary Claim Constructions And Extrinsic Evidence Pursuant To Patent L.R. 4-2 and further discussed below.

4. This declaration sets forth my opinion regarding the meaning of disputed terms in the Asserted Claims and describes how a person of ordinary skill in the art would have understood those terms at the filing date of the applications to which the '769 Patent and the '903 Patent Family, respectively, claim priority. In addition, this declaration provides an overview of the relevant technology.

A. Qualifications

5. Since 2004, I have been a Professor of Music and an Associate Professor (by courtesy) of Electrical Engineering at Stanford University, based at the Center for Computer Research in Music and Acoustics (CCRMA) in Stanford, California. From 1994 to 2004, I was an Associate Professor of Music and (by courtesy) Electrical Engineering at Stanford University, at CCRMA. Before that, from 1989 to 1994, I was an Associate Professor (Research) at CCRMA, with my research focusing on signal processing techniques applied to music and audio. I also taught signal processing courses in support of my research program, and supervised Ph.D. research in the Computer Music Ph.D. program at CCRMA and in the Electrical Engineering department.

6. My research since the 1970s has focused on advanced signal processing methods applied primarily to audio signals. Among other signal processing topics, I have studied and employed Time Scale Modification (TSM) technology both in my teaching and in my consulting work. In particular, I have implemented Synchronous OverLap Add

(SOLA) methods for TSM, and have described and demonstrated SOLA methods in my teaching. One of the books I have authored, *Spectral Audio Signal Processing*, includes a section on TSM methods. The audio signal processing technologies I have mastered, implemented, and taught encompass the technologies used in TSM in both the time and frequency domains.

7. I teach both undergraduate and graduate courses focusing on signal processing and music technology at Stanford University. Specific courses include "Introduction to Digital Audio Signal Processing," "Signal Processing Models in Musical Acoustics," "Software for Sound Synthesis and Audio Effects," "Audio Applications of the Short-Time Fourier Transform," and "Projects in Spectral Audio Signal Processing." These courses were developed starting in 1984 and I have taught variants of them every year since 1984 except when on sabbatical leave.

8. I have published many scholarly articles and papers in peer-reviewed journals and conferences in the fields of signal processing, computer music, acoustics, speech, and electrical engineering. At the time of this writing, I had published a total of 250 peer-reviewed journal articles and conference papers. All of these publications are related to the fields of signal processing, computer music, acoustics, speech, and electrical engineering.

9. I have also published several books. I was the sole author of the following books: *Mathematics of the Discrete Fourier Transform (DFT) with Audio Applications*, published by W3K Publishing in 2007, *Introduction to Digital Filters with Audio Applications*, published by W3K Publishing in 2007, *Physical Audio Signal Processing*, published by W3K Publishing in 2010, and *Spectral Audio Signal Processing*, published by W3K Publishing in 2011.

10. I received a Bachelor of Science in Electrical Engineering degree in Control Communications and Circuits from Rice University in 1975. I received a Master of Science in Electrical Engineering degree with concentration in Statistical Signal Processing and a Doctor of Philosophy (Ph.D.) in Electrical Engineering from Stanford University in 1978 and 1983, respectively.

11. From the fall of 1982 until 1986, I worked half-time for Systems Control Technology in Palo Alto, CA, on projects that included time-delay estimation, autoregressive-moving-average (ARMA) modeling and spectrum analysis, underwater acoustic signal processing, high frequency (HF) communications signal processing, and general tool development.

12. During the time period from 1984 to 1986, I also worked half-time as a research associate at Stanford University, at CCRMA. My research focused on violin modeling, woodwind modeling, new digital filter design methods tailored to audio applications, new reverberation techniques, time-varying sampling-rate conversion, digital filtering, software, spectrum analysis software, system identification software, and pitch detection. Duties included teaching a two-year sequence in digital signal processing aimed at graduate students interested in music applications of signal processing and acoustics.

13. Between 1986 and 1993, I worked as a software engineer at NeXT Inc., and was responsible for signal processing software pertaining to music and audio. I designed and implemented a variable-rate transform coder for real-time compression/decompression of "CD-quality" audio signals. I also managed the Sound, Music, and Signal Processing Group at NeXT from its inception in 1986. I co-designed the NeXT Music Kit, wrote and supported the NeXT DSP Library, and helped support and debug the Sound/DSP Mach driver and the NeXT Sound Library.

14. I am being compensated for my time spent on the case at a rate of $375/hour. My compensation is not dependent on the outcome of this case.

15. I attach as Exhibit A my updated curriculum vitae setting forth my qualifications and publications.

B. Materials Reviewed

16. For the purposes of this declaration, I have reviewed the following materials:

- The '769 Patent;
- The prosecution history of the '769 Patent;
- The '903 Patent;
- The prosecution history of the '903 Patent;
- The '050 Patent;
- The prosecution history of the '050 Patent;
- The '720 Patent;
- The prosecution history of the '720 Patent;
- Apple's Proposed Claim Terms for Construction Pursuant to Patent L.R. 4-1;
- EPL's Proposed Claim Terms for Construction Pursuant to Patent L.R. 4-1;
- Apple Inc.'s Exchange Of Preliminary Claim Constructions And Extrinsic Evidence Pursuant To Patent L.R. 4-2;
- EPL Holdings, LLC's Exchange Of Preliminary Claim Constructions And Extrinsic Evidence;
- Plaintiff EPL Holding, LLC's Amended Disclosure Of Asserted Claims And Infringement Contentions;

- Hejna, Donald J., "Real-Time Time-Scale Modification of Speech via the Synchronized Overlap-Add Algorithm" (Thesis), Massachusetts Institute of Technology;
- Hejna and Musicus, "The SOLAFS Time-Scale Modification Algorithm," *BBN*, July 1991;
- Roucos and Wilgus, "High Quality Time-Scale Modification for Speech," *Acoustics, Speech, and Signal Processing, IEEE International Conference on ICASSP*, Vol. 10 (1985);
- Schwartzman, *The Words of Mathematics: An Etymological Dictionary of Mathematical Terms Used in English* (The Mathematical Assoc. of Am., 1996);
- The New Oxford American Dictionary, 2001;
- The American Heritage Dictionary, 2nd College Edition, 1991;
- The Oxford English Dictionary, 2nd Edition, 1989.
- Smith, Julius O., *Spectral Audio Signal Processing*, W3K Publishing, 2011.

II. UNDERSTANDING OF THE LAW

17. I am not an attorney. For the purposes of this declaration, I have been informed about certain aspects of the law that are relevant to my analysis and opinions.

18. I have been informed and understand that before an infringement or validity determination can be made, the claims must be construed. I understand that "claim construction" is the process of determining a patent claim's meaning. I also have been informed and understand that the proper construction of a claim term is the meaning that a person of ordinary skill in the art (i.e., the technical field to which the patent relates) would have given to that term at the time when the patent's application was filed.

19. I have been informed and understand that a claim construction analysis begins with the ordinary meaning of the disputed claim term, and there is a presumption that claim terms carry their accustomed meaning among persons of ordinary skill in the art. I have also been informed that the ordinary and customary meaning of a claim term may be determined by reviewing a variety of sources, including the claims themselves, the specification (or "written description") of a patent, its prosecution history, and dictionaries and treatises.

20. I understand that, for claim construction, one must focus on the claim terms in the context of the claim as a whole, interpreting the claim language as it ordinarily would be understood. After the claim language, the most important sources to consider are the patent specification, including any publications incorporated by reference in the

specification, and the prosecution history. I understand that, collectively, these sources—the claim language, specification, and prosecution history—are called "intrinsic evidence."

21. With respect to technical references, and other information—called "extrinsic evidence"—that would have been available at the time when the patent's application was filed, I understand that the law considers extrinsic evidence to be less reliable than intrinsic evidence, and that extrinsic evidence cannot change the ordinary meaning of the claim language. I understand that although certain terms may be at the center of the claim construction debate, the context of the surrounding words of the claim also must be considered in determining the ordinary and customary meaning of those terms. I also understand that limitations generally may not be imported from the specification into the claims.

22. It is my understanding that, after ascertaining the ordinary meaning of the claim terms, the specification and prosecution history must be consulted to determine whether the presumption of ordinary meaning has been rebutted. I understand that a patentee may act as his own lexicographer and provide definitions in the specification or prosecution history. For example, an inventor may clearly set forth a definition of the disputed claim term in either the specification or prosecution history that is different from the term's ordinary meaning. I have been informed that there may also be a clear disavowal of claim coverage in the specification or prosecution history, and that in some circumstances, the ordinary meaning of the claim terms may deprive the claim of clarity so that resort to other intrinsic evidence for a definite meaning is required.

23. I understand that actions taken by the patent owner can affect the constructions of the claim terms. In particular, if the patent owner distinguishes the claims from prior art during prosecution, it would be incorrect to construe the claims so as to cover the distinguished material. Similarly, I understand that a patentee's decision to narrow his or her claims through amendment is presumed to be a general disclaimer of the territory between the original claim and the amended claim.

III. PERSON OF ORDINARY SKILL IN THE ART

24. The application that issued as the '769 Patent was filed in the U.S. Patent and Trademark Office on July 23, 1991 and does not claim priority to any earlier filing. Accordingly, I understand that I am to interpret the claim terms in the '769 Patent from the perspective of the person of ordinary skill in the art as of July 23, 1991.

25. Based on my understanding of the '769 Patent and my education, knowledge, and industrial and academic experience, a person of ordinary skill in the art of the '769 Patent would have a bachelor's degree in electrical engineering or computer science, or its equivalent, and two years of experience related to time-scale modification algorithms.

26. The application that issued as the '903 Patent was filed in the U.S. Patent and Trademark Office on August 17, 2004, and on its face claims priority to an application filed on December 11, 2001.

27. The application that issued as the ’050 Patent was filed in the U.S. Patent and Trademark Office on April 24, 2012, and on its face claims priority to a provisional application filed on December 12, 2000.

28. The application that issued as the ’720 Patent was filed in the U.S. Patent and Trademark Office on November 23, 2011, and on its face claims priority to a provisional application filed on December 12, 2000.

29. Accordingly, I understand that I am to interpret the claim terms in the ’903 Patent Family from the perspective of the person of ordinary skill in the art as of December 12, 2000. Notably, even if December 11, 2001 is the appropriate date for the ’903 Patent, it would not change my analysis.

30. Based on my understanding of the ’903 Patent Family and my education, knowledge, and industrial and academic experience, a person of ordinary skill in the art of the ’903 Patent Family would have a bachelor’s degree in electrical engineering or computer science, and two years of programming experience related to media playback software.

IV. THE ’769 PATENT DISCLOSURE

31. The ’769 patent issued on December 29, 1992, and is directed to “time-scale modification (‘TSM’) of a signal,” i.e., audio playback at rates that are faster or slower than the default audio speed without altering the pitch of the audio. (’769 patent at 1:6-13.) In particular, the ’769 patent claims a method for “changing the perceived rate of articulation while ensuring that the local pitch period of the resulting signal remains unchanged, i.e., there are no ‘Alvin the Chipmunk’ effects.” (Id. at 1:21-26.)

32. According to the ’769 patent, a variety of algorithms for speeding or slowing audio playback without changing the perceived pitch were known in the art, including “frequency domain processing methods, analysis/synthesis methods, and time-domain processing methods.” (*Id.* at 1:63-67.) The ’769 patent does not claim to disclose a new type of TSM algorithm, but instead claims a purportedly “inventive method [that] is an improvement on the SOLA method described in the Background of the Invention.” (*Id.* at 4:57-61.) The referenced prior art SOLA (“Synchronized Overlap-Add”) method was reported in technical literature published in the 1980s, and was not discovered by the named inventors of the ’769 patent. (*Id.* at 3:5-12; Roucos and Wilgus, “High Quality Time-Scale Modification for Speech,” *Acoustics, Speech, and Signal Processing, IEEE International Conference on ICASSP*, Vol. 10 (1985).) The ’769 patent notes that the prior art SOLA method is advantageous in that it “has low complexity and . . . operates without regard to pitch periods in a speech signal.” (’769 patent at 3:8-12.) The SOLA algorithm “modifies the time-scale of an input signal in two steps which are referred to as analysis and synthesis, respectively.” (*Id.* at 3:48-51.) In the “analysis step,” the audio signal is divided into “input window[s] ha[ving] a fixed length W and separated by a fixed analysis distance”; in the “synthesis step” “[t]he windows [from the analysis step] are overlap-added at a dynamic synthesis rate.” (*Id.* at 3:17-18, 3:48-56.) Because the second “synthesis step” of the SOLA method is “dynamic,” “[e]ach new window is

aligned with" the output signal "before being added to reduce discontinuities in the resulting signal." (*Id.* at 3:64-4:4.) This *alignment* is what distinguishes *synchronous* overlap-add (SOLA) from regular overlap-add (OLA).

33. The '769 patent describes what it contends were drawbacks of the prior art SOLA algorithm, and attempts to set forth an incremental improvement upon that prior art TSM method. According to the patent, SOLA's first drawback was that calculating the amount of overlap for each frame from the input signal with the time-compressed output signal "complicates the work required to compute the similarity measure and to fade across the overlap region." (*Id.* at 4:36-41.) The second drawback, the '769 patent contends, is that "more than two windows may overlap in certain regions and this further complicates the fading computation." (*Id.* at 4:41-43.) In other words, according to the '769 patent, it is simpler to measure the similarity of a fixed overlap-region in the output signal to a range of considered locations in the input signal.

34. The '769 invention purports to solve these problems by disclosing an "inventive method [that] is an improvement on the SOLA method described in the Background of the Invention and is referred to here as a Synchronized Overlap-Add, Fixed Synthesis time domain processing method ('SOLAFS')." (*Id.* at 4:57-61.) In the SOLAFS algorithm, the '769 inventors shifted the "dynamic" aspect of the SOLA algorithm from the second synthesis step to the first analysis step. Thus, where the first "analysis step" of the prior art SOLA algorithm divides the signal to be modified into "window[s] of a <u>fixed</u> length W . . . separated by a <u>fixed</u> distance," "[in] accordance with the ['769] invention, blocks of the input signal, referred to as analysis windows, are taken at an average rate . . . with <u>each starting position allowed to vary</u> within limits." (*Id.* at 3:54-56, 5:7-13 (emphasis added).) Likewise, where the second step of the prior art SOLA algorithm "overlap-add[s the windows] at a <u>dynamic</u> synthesis rate to provide an output," in the '769 "Synchronized Overlap-Add <u>Fixed Synthesis</u>" algorithm, the "output signal is reconstructed using a <u>fixed</u> inter-block offset S_s, i.e., the duration of overlap with the existing signal in each window to be added is fixed." (*Id.* at 3:17-18, 5:7-13 (emphasis added).) SOLA employs a *fixed analysis rate* to accomplish TSM, whereas SOLAFS instead uses a *fixed synthesis rate.* Similarly, SOLA uses a *dynamic synthesis rate* to accomplish TSM, while SOLAFS uses a *dynamic analysis rate* to accomplish TSM. The following table summarizes the fixed and dynamic steps for each of the SOLA and SOLAFS algorithms:

	SOLA (prior art)	**SOLAFS ('769 patent)**
Analysis Step	Fixed	Dynamic
Synthesis Step	Dynamic	Fixed

In other respects, SOLA and SOLAFS share the same elementary signal processing techniques that were well known in the art such as windowing, overlap-add, cross-correlation, and time alignment, to name a few.

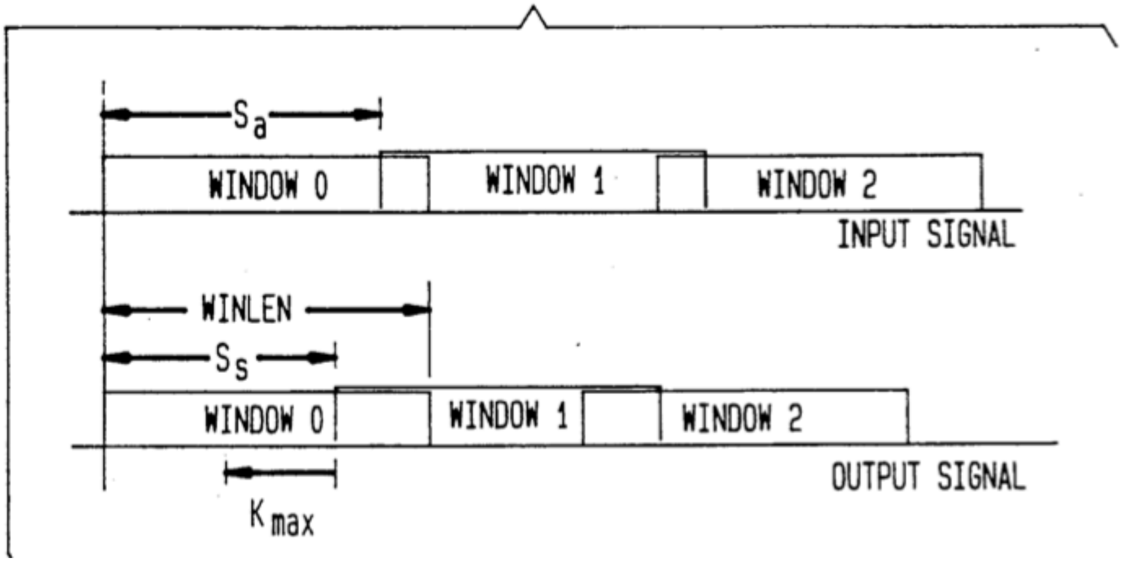
35. In disclosing its purported invention, the ’769 patent adopts the nomenclature of the prior art SOLA method that it seeks to improve, and retains each parameter utilized by the prior art SOLA method, including W, S_s, S_a, K_{max}, and W_{ov}.

> The inventive method is an improvement of the prior SOLA method discussed in the Background of the Invention, which inventive method is referred to as the Synchronized Overlap-Add, Fixed Synthesis method (“SOLAFS”). With reference to FIGS. 1 and 2, there are four parameters which are used in the inventive SOLAFS method: (a) window length W is the duration of windowed segments of the input signal--this parameter is the same for input and output buffers and represents the smallest unit of the input signal, for example, speech, that is manipulated by the method; (b) analysis shift S_a is the interframe interval between successive search ranges for analysis windows along the input signal; (c) synthesis shift S_s is the interframe interval between successive analysis windows along the output signal; and (d) shift search interval K_{max} is the duration of the interval over which an analysis window may be shifted for purposes of aligning it with the region of the output signal it will overlap.
>
> In essence, the first W_{OV} samples in each new window in the input signal, referred to as an analysis window, are overlap-added with the last W_{OV} samples in the output signal, i.e., this is referred to as overlap-adding at a fixed synthesis rate. In accordance with the inventive method, the starting point of each analysis window is varied by: (a) evaluating a similarity measure such as, for example, the cross-correlation, of the first W_{OV} points in the analysis window with the last W_{OV} points in the output signal, where W_{OV} is a predetermined, fixed number; (b) then the starting point of the analysis window is shifted by a fixed amount and a new cross-correlation of the first W_{OV} points in the new analysis window with the same last W_{OV} points in the output signal is evaluated; (c) step (b) is performed a predetermined number of times, K_{max}, and the new analysis window is chosen to be the one wherein the cross-correlation is maximized. Finally, the first W_{OV} samples in the new analysis window are overlap-added with the last W_{OV} samples in the output signal and S_s additional points from the analysis window are appended to the output signal The term overlap-added refers to a method of combination such as averaging points or performing a weighted average in accordance with a predetermined weighting function.

(*Id.* at 7:3-47.)

36. Figure 1 of the ’769 patent indicates the use of these parameters to divide the input signal into overlapping “frames” or “windows,” change the spacing in between them to alter the rate of playback and overlap them, and to optimize the alignment of the “frames” or “windows” over the interval K_{max}.

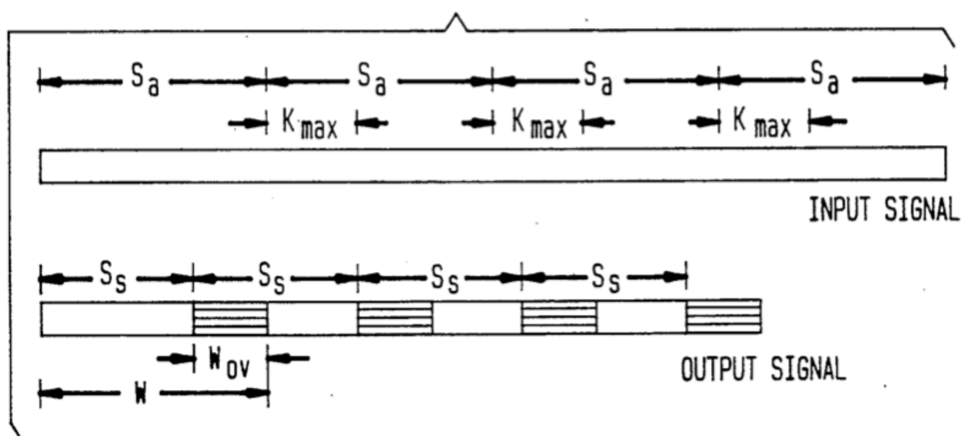
FIG. 1
(PRIOR ART)



(*Id.* at FIG. 1.)

37. Figure 2 of the ’769 patent pertains to the purported invention, and also indicates the use of the prior art SOLA method parameters to modify the time-scale of audio playback. The only significant difference between Figure 1 (prior art SOLA method) and the Figure 2 (purported ’769 invention) is that K_{max} – the interval over which the “frames” or “windows” may be shifted to optimize their alignment – has been moved from the output signal (prior art SOLA method) to the input signal (purported ’769 invention).

FIG. 2



(*Id.* at FIG. 2.)

V. THE ’769 PATENT CLAIMS AND FILE HISTORY

38. ’769 Asserted Claims 1, 2, 10, 11, and 19 correspond to original claims 1, 2, 10, 11, and 19, respectively, in U.S. Application Serial No. 734,424. The claims are reproduced below, with disputed claim terms indicated in **bold**:

> 1. A method for **time-scale modification** of a signal comprised of an input stream of signal representations to form an output stream of signal representations, the method comprising the steps of:
>
> **determining an input block of W signal representations from the input stream** for use in overlapping signal representations from the input block with signal representations in the output stream; and
>
> overlapping $\mathbf{W_{OV}}$ signal representations from the beginning of the input block with $\mathbf{W_{OV}}$ signal representations from the end of the output stream, where $\mathbf{W_{OV}}$ is **determined by W** and the **time-scale modification**.
>
> 2. The method of claim 1 wherein the step of overlapping comprises the step of:
>
> applying a weighting function to $\mathbf{W_{OV}}$ signal representations from the beginning of the input block and to $\mathbf{W_{OV}}$ signal representations from the end of the output stream to determine values of $\mathbf{W_{OV}}$ signal representations to be substituted for the $\mathbf{W_{OV}}$ signal representations at the end of the output stream; and wherein the step of overlapping further comprises the step of:
>
> placing $\mathbf{W - W_{OV} = S_s}$ signal representations from the input stream at the end of the output stream, the S_s signal representations being subsequent to the $\mathbf{W_{OV}}$ signal representations from the beginning of the input block.
>
> 10. A method for **time-scale modification** of a signal comprised of an input stream of signal representations to form an output stream of signal representations, the method comprising the steps of:
>
> determining a number of signal representations for use in overlapping signal representations from the input stream to the output stream, $\mathbf{W_{OV}}$;
>
> **determining an input block of W signal representations from the input stream** for use in overlapping signal representations from the input block with signal representations in the output stream; and

> overlapping $\mathbf{W_{OV}}$ signal representations from the beginning of the input block with $\mathbf{W_{OV}}$ signal representations from the end of the output stream.
>
> 11. The method of claim 10 wherein the step of overlapping comprises the step of:
>
> applying a weighting function to $\mathbf{W_{OV}}$ signal representations from the beginning of the input block and to $\mathbf{W_{OV}}$ signal representations from the end of the output stream to determine values of $\mathbf{W_{OV}}$ signal representations to be substituted for the $\mathbf{W_{OV}}$ signal representations at the end of the output stream; and wherein the step of overlapping further comprises the step of:
>
> placing $\mathbf{W} - \mathbf{W_{OV}} = \mathbf{S_s}$ signal representations from the input stream at the end of the output stream, the S_s signal representations being subsequent to the $\mathbf{W_{OV}}$ signal representations from the beginning of the input block.
>
> 19. A method which comprises the steps of:
>
> **time-scale modifying** a signal comprised of an input stream of signal representations to form an output stream of signal representations wherein at least one of the steps of time-scale modifying comprises:
>
> **determining an input block of signal representations from the input stream** for use in appending signal representations from the input block to signal representations in the output stream, where the number appended is **determined by** the time-scale modification; and
>
> appending the signal representations to the end of the output stream.

39. On August 20, 1992, in response to the applicant's original patent application, the PTO mailed a Notice of Allowability allowing claims 1-20. These claims issued as claims 1-20 of the '769 Patent. In this action, the examiner stated:

> "EXAMINER REASONS FOR ALLOWANCE; the claimed: 'Time-scale modifying a signal . . . determining an input block for use in overlapping (or appending) . . .' further limited by the other elements of the claim, is not suggested by the prior art."

VI. DISPUTED CLAIM TERMS OF THE '769 PATENT

40. The following terms are disputed between the parties:

A. "determining an input block of W signal representations from the input stream" / "determining an input block of signal representations from the input stream"

Claim Nos.	Proposed Term	Apple's Proposed Construction	EPL's Proposed Construction
1, 10	determining an input block of W signal representations from the input stream / determining an input block of signal representations from the input stream	"searching for and identifying the starting position of an input block of W signal representations that is similar to the output stream" / "searching for and identifying the starting position of an input block of signal representations that is similar to the output stream"	*Plain and Ordinary Meaning subject to the term "W" defined below*

41. A person of ordinary skill in the art reading the term "determining an input block of W signal representations from the input stream" in the context of the asserted '769 claims would understand this term to mean "searching for and identifying the starting position of an input block of W signal representations that is similar to the output stream."

42. Likewise, a person of ordinary skill in the art reading the term "determining an input block of signal representations from the input stream" in the context of the asserted '769 claims would understand this term to mean "searching for and identifying the starting position of an input block of signal representations that is similar to the output stream."

1. The '769 Patent Claims, Specification, and Figures

43. A key feature of the purported invention is searching for and identifying the starting position of an input block of signal representations that is similar to the output stream. A person of ordinary skill in the art reading the Asserted Claims would understand by their context that the disputed terms pertain to this aspect of the purported '769 invention.

44. For example, in claim 1 of the '769 patent, the disputed term pertains to only one of two limitations describing the steps of the purportedly inventive time-scale modification method.

> 1. A method for time-scale modification of a signal comprised of an input stream of signal representations to form an output stream of signal representations, the method comprising the steps of:

> **determining an input block of W signal representations from the input stream** for use in overlapping signal representations from the input block with signal representations in the output stream; and
>
> overlapping W_{OV} signal representations from the beginning of the input block with W_{OV} signal representations from the end of the output stream, where W_{OV} is determined by W and the time-scale modification.

The disputed terms appear in a similar context in independent claims 10 and 19. Accordingly, a person of ordinary skill in the art reading the disputed terms in the context of the Asserted Claims would understand them to mean "searching for and identifying the starting position of an input block of [W] signal representations that is similar to the output stream."

45. The "SUMMARY OF THE INVENTION" section of the '769 specification emphasizes that searching for and identifying the starting position of an input block that is similar to the output stream is a key feature of the purported invention:

> The inventive method is an improvement on the SOLA method described in the Background of the Invention and is referred to here as a Synchronized Overlap-Add, Fixed Synthesis time domain processing method ("SOLAFS"). ***In general, the inventive method comprises superimposing partially overlapping blocks of signal samples from an input signal in a manner which aligns similar signal blocks from different locations in the input signal.***

(*Id.* at 4:57-65 (emphasis added).)

46. The "SUMMARY OF THE INVENTION" section provides specific detail as to how the '769 patent's algorithm searches for and identifies the starting position of an input block that is similar to the output stream:

> In accordance with the present invention, blocks of the input signal, referred to as analysis windows, are taken at an average rate of S_a ***with each starting position allowed to vary within limits*** and an output signal is reconstructed using a fixed inter-block offset S_s, i.e., ***the duration of overlap with the existing signal in each window to be added is fixed***. ***This is done by searching for segments of the input signal near the target starting position*** mS_a ***which are similar to the portion of the output signal that will overlap when constructing the output signal. A similarity measure is used to evaluate such similarity and, in accordance with the present invention, the similarity measure uses a fixed, predetermined minimum number of samples***. . . . Several similarity measures are evaluated by shifting the starting point of an analysis window over a predetermined number of samples, i.e., removing samples from the beginning of the analysis window as new samples from the input are appended to the tail of the analysis window, thus using the same,

> predetermined number of samples in the evaluation. ***The starting position of the analysis window which provides the maximum similarity in the region of the analysis window which will overlap with the region of the output signal is selected from all starting positions tested***. Finally, the predetermined number of samples in the region of overlap are combined with the predetermined number of samples from the end of the previous portion of the output signal and the remaining samples in the window are appended to the combined segment of the previous portion of the output signal.

(*Id.* at 5:7-41 (emphasis added).)

47. Likewise, the "DETAILED DESCRIPTION" section of the '769 specification explains that the '769 patent's algorithm searches for and identifies the starting position of an input block that is similar to the output stream. Notably, the '769 specification adopts the nomenclature for the parameters of the SOLA algorithm (discussed below) for its description of the purportedly inventive SOLAFS algorithm:

> The inventive method is an improvement of the prior SOLA method discussed in the Background of the Invention, which inventive method is referred to as the Synchronized Overlap-Add, Fixed Synthesis method ("SOLAFS"). ***With reference to FIGS. 1 and 2, there are four parameters which are used in the inventive SOLAFS method***: (a) window length W is the duration of windowed segments of the input signal--this parameter is the same for input and output buffers and represents the smallest unit of the input signal, for example, speech, that is manipulated by the method; (b) analysis shift S_a is the interframe interval between successive search ranges for analysis windows along the input signal; (c) synthesis shift S_s is the interframe interval between successive analysis windows along the output signal; and (d) ***shift search interval K_{max} is the duration of the interval over which an analysis window may be shifted for purposes of aligning it with the region of the output signal it will overlap***.
>
> In essence, the first W_{OV} samples in each new window in the input signal, referred to as an analysis window, are overlap-added with the last W_{OV} samples in the output signal, i.e., this is referred to as overlap-adding at a fixed synthesis rate. ***In accordance with the inventive method, the starting point of each analysis window is varied by: (a) evaluating a similarity measure such as, for example, the cross-correlation, of the first W_{OV} points in the analysis window with the last W_{OV} points in the output signal, where W_{OV} is a predetermined, fixed number; (b) then the starting point of the analysis window is shifted by a fixed amount and a new cross-correlation of the first W_{OV} points in the new analysis window with the same last W_{OV} points in the output signal is evaluated; (c) step (b) is performed a predetermined number of times, K_{max}, and the new analysis window is chosen to be the one wherein the cross-correlation is***

> ***maximized.*** Finally, the first W_{OV} samples in the new analysis window are overlap-added with the last W_{OV} samples in the output signal and S_s additional points from the analysis window are appended to the output signal The term overlap-added refers to a method of combination such as averaging points or performing a weighted average in accordance with a predetermined weighting function.

(*Id.* at 7:3-47 (emphasis added).)

FIG. 2

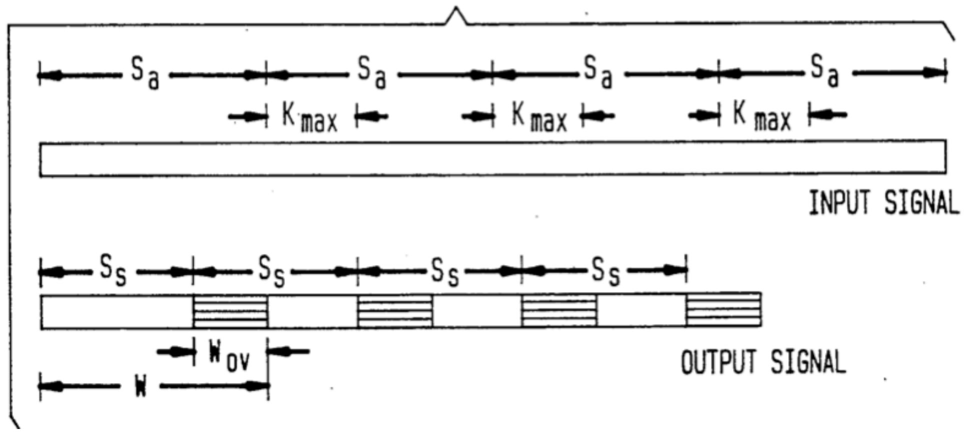
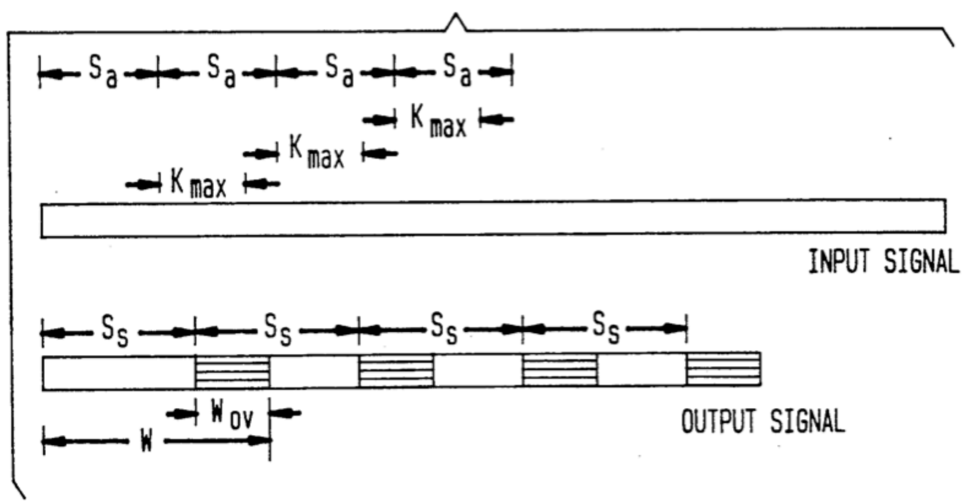


FIG. 3



(*Id.* at FIGS. 2 and 3.) Notably, as set forth above, the SOLAFS algorithm adopts the parameter $\mathbf{K_{max}}$ as "the duration of the interval over which an analysis window may be shifted for purposes of aligning it with the region of the output signal it will overlap." The inclusion of $\mathbf{K_{max}}$ in the SOLAFS algorithm indicates the interval over which the method searches for and identifies the starting position of an input block that is similar to the output stream.

48. The '769 patent also contends that the dynamic synthesis aspect of the prior art SOLA method is unnecessarily computationally intensive:

> The SOLA method has a drawback in that the amount of overlap for the m^{th} window, W^m_{OV}, between the output and the m^{th} analysis window varies with $\mathbf{k_m}$ and ***this complicates the work required to compute the similarity measure and to fade across the overlap region***. Also, depending on the shifts $\mathbf{k_m}$, more than two windows may overlap in certain regions and this further complicates the fading computation.

(*Id.* at 4:36-43 (emphasis added).)

49. The '769 patent purports to reduce the computational demands of the dynamic synthesis of the prior art SOLA method. However, the '769 patent nowhere discloses nor suggests reducing the computational demands of its purportedly inventive time-scale modification algorithm by foregoing synchronization. Instead, the '769 inventors suggested that their SOLAFS algorithm would "improve" on the SOLA algorithm by switching the dynamic search that maximizes a similarity measure from the later synthesis step to the earlier analysis step. For example, as stated in the '769 abstract:

> Method for time-scale modification ("TSM") of a signal, for example, a voice signal, ***wherein starting positions of blocks in an input signal, referred to as analysis windows, are varied*** and an output signal is reconstructed by overlapping analysis windows using fixed window offsets, i.e., the duration of overlap between analysis windows is fixed during reconstruction. ***This is done by searching for segments of the input signal which are similar to the previous portion of the output signal.*** In one embodiment of the present invention a cross-correlation is used as a similarity measure to evaluate such similarity and the cross-correlation uses a fixed, predetermined minimum number of samples. ***The starting position of the analysis window which results in the greatest similarity in overlapping regions is determined as the starting position which provides the largest value of cross-correlation in the overlapping regions***. ***Several cross-correlations are evaluated by shifting the analysis window over a predetermined number of samples, removing the first shifted samples in the evaluation each time, and using the same, predetermined number of samples in the evaluation to determine the "best" starting position for an analysis window***.

('769 patent at abstract (emphasis added).)

50. The mathematical equations describing the '769 patent's algorithm likewise confirm that the SOLAFS algorithm searches for and identifies the starting position of an input block that is similar to the output stream. In particular, those equations detail that, in the SOLAFS method, the input windows are shifted a number $\mathbf{k_m}$ of shifts during the analysis phase, followed by fixed synthesis of the output signal:

In the following x[i] represents the ith sample in the input digital stream representative of an input signal. In accordance with the inventive method, analysis windows are chosen as follows:

$$x_m[n] = \begin{cases} x_m[mS_a + k_m + n] & \text{for } n = 0, \ldots, W-1 \\ 0 & \text{otherwise} \end{cases}$$

where: m is a window index, i.e., it refers to the m^{th} window; n is a sample index in an input buffer for the input signal, which buffer is W samples long; ***k_m is the number of samples of shift for the m^{th} window***; and $x_m[n]$ represents the n^{th} sample in the m^{th} analysis window.

The analysis windows are then used to form the output signal y[i] recursively in accordance with the following:

$$y[mS_s+n] \leftarrow b[n]y[mS_s+n]+(1-b[n])x_m[n] \text{ for } n=0, \ldots, W_{OV}-1 \quad (5)$$

and

$$y[mS_s+n] \leftarrow x_m[n] \text{ for } n=W_{OV}, \ldots, W-1 \quad (6)$$

where: W_{OV} =W-S_s is the number of points in the overlap region and b[n] is an overlap-add weighting function which is referred to as a fading factor--an averaging function, a linear fade function, and so forth.

Note that, in accordance with the present invention, shift k_m affects the starting position of an analysis window in the input digital stream. For a particular window, an optimal shift is determined by maximizing a similarity measure between the overlapping samples in x_m and y.

(*Id.* at 7:48 – 8:15 (emphasis added).)

51. The '769 patent further describes specific examples in connection with Fig. 4 that search for and identify the starting position of an input block that is similar to the output stream. (*Id.* at 8:67 – 10:20; FIG. 4.) For example, the cited description states:

[W]e will find the maximum similarity between the first W_{OV} samples, i.e., 2 samples in this case, at the start of the analysis window and the end of the output signal. ***Referring to line 102 of FIG. 4, we compute the cross-correlation between samples 5 and 6 from the start of the analysis window and samples 2 and 3 from the end of the output window. Next,***

> ***we shift the start of the analysis window by one and repeat the process.*** This is indicated as line 103 in FIG. 4 where we compute the cross-correlation between samples 6 and 7 from the new start of the analysis window and samples 2 and 3 from the end of the output window. This process is continued until we have shifted the analysis window by a maximum amount K_{max} which is allowed. Then, ***we determine which shift corresponds to the maximum cross-correlation***.

(*Id.* at 9:11-26 (emphasis added).)

> [W]e will find the maximum similarity between the first W_{OV} samples, i.e., 2 samples in this case, at the start of the analysis window and the end of the output signal. Referring to line 202 of FIG. 4, ***we compute the cross-correlation between samples 2 and 3 from the start of the analysis window and samples 3 and 4 from the end of the output window. Next, we shift the start of the analysis window by one and repeat the process*** This is indicated as line 203 in FIG. 4 where we compute the cross-correlation between samples 3 and 4 from the new start of the analysis window and samples 3 and 4 from the end of the output window This process is continued until we have shifted the signal by the maximum amount K_{max} which is allowed. Then, ***we determine which shift corresponds to the maximum cross-correlation***.

(*Id.* at 9:46-61.)

52. The ’769 patent details an example in connection with Fig. 8 that searches for and identifies the starting position of an input block that is similar to the output stream. (*See id.* at 10:66-11:62.) Likewise, the ’769 patent’s description in connection with Figs. 5-7 discloses searching for and identifying the starting position of an input block that is similar to the output stream:

> FIGS. 5-7 show a flowchart of one embodiment of the inventive SOLAFS method. The following is nomenclature which is used in the following flowchart: (a) W is the window length and represents the smallest block or unit of a signal that is manipulated by the inventive method; (b) S_a is the analysis shift and represents the interframe interval between successive search intervals along the input signal; (c) S_s is the synthesis shift and represents the interframe interval between successive windows in the output signal; (d) ***k_m is the window shift and represents the number of data samples the m^{th} analysis window is shifted from its target position, mS_a, to provide alignment with previous windows***; (e) ***K_{max} is the maximum window shift, i.e., $0 \leq k_m \leq K_{max}$ for all m***; (f) W_{OV} =W-S_s is the fixed number of overlapping points between windows; (g) ***head_buf is a storage buffer for samples from an input signal buffer, head_buf has a length of K_{max} +W***; and (h) tail_buf is a storage buffer of length W_{OV}.

(*Id.* at 12:41-60 (emphasis added); *see also id.* at 12:61 – 14:21).)

53. Significantly, the '769 specification acknowledges that the bulk of the computational requirements with its SOLAFS method stem from searching for and identifying the starting position of an input block that is similar to the output stream. (*See, e.g., id.* at 11:63-65 ("The bulk of the computation in the inventive SOLAFS method revolves around computing the normalized cross-correlation R^{m}_{xy} [k] and choosing the maximum[.]").) However, despite considering multiple possibilities for reducing those computational demands, the '769 nowhere suggests that the SOLAFS algorithm could be simplified by foregoing searching for and identifying the starting position of an input block that is similar to the output stream. (*See, e.g. id.* at 11:63 – 12:29; 14:34-45; 15:41 – 16:32.)

54. In addition to the above-discussed citations, the '769 patent expressly discloses that the purportedly inventive SOLAFS method must include searching for and identifying the starting position of an input block that is similar to the output stream. For example, the '769 patent discloses that the SOLAFS method "requires" a W+$\mathbf{K_{max}}$ length buffer to operate:

> "***The inventive SOLAFS method requires*** a W_{OV} length output buffer to hold the last samples of the output, i.e., y[mS_s], . . . , y[mS_a +W_{OV} -1], and ***a W+K_{max} length input buffer to hold the input samples that might be used in the next analysis window***, x[mS_a], . . . , x[mS_a +W+K_{max} -1]. One must take note of the fact that in a real-time application, time-scale compression will require reading in input data at a much faster rate than usual. This may cause difficulties if the data is stored in compressed form and must be decoded, or if the storage unit is slow."

(*Id.* at 12:30-40 (emphasis added).) If the claimed invention did not require searching for and identifying the starting position of an input block of W signal representations that is similar to the output stream, it would not "require[]" a W+$\mathbf{K_{max}}$ length buffer to operate.

55. Likewise, the '769 patent discloses that $\mathbf{K_{max}}$ – the interval over which the SOLAFS method searches for and identifies the starting position of an input block that is similar to the output stream – "must" be larger than the expected pitch period of the input signal. In other words, $\mathbf{K_{max}}$ must be significantly larger than zero.

> The ***maximum shift K_{max}*** is an important parameter. ***This must be chosen to be larger than the largest expected pitch period in the input signal to avoid pitch fracturing***. In a voicemail application with male speakers and 8 kHz sampling, a preferred choice is $\mathbf{K_{max}}$ =100 samples. This choice allows synchronization of periods down to 80 Hz when time-scale modifying music as well.

(*Id.* at 14:63 – 15:2 (emphasis added); *see also id.* at 15:11-31 (disclosing "parameter choices provid[ing] high-quality output" where $\mathbf{K_{max}}$ = 100); *see also id.* at 16:33-47 ("We have determined that ***alignment is most critical*** during voiced portions of speech signals.").) *See also*:

> "Although ***the inventive SOLAFS method*** has been described with reference to the application thereof to samples of a signal for ease of understanding, it should be noted that the inventive method is not limited to operating on samples of the signal. ***In particular, the method operates by searching for similar regions in an input and an output and then overlapping the regions to produce a time-scale modified output***."

(*Id.* at 16:63 – 17:2.) A person of ordinary skill in the art reading these disclosures would therefore understand the disputed terms in the Asserted Claims to require "searching for and identifying the starting position of an input block of [W] signal representations that is similar to the output stream."

56. Consistent with the above-described disclosures, the '769 patent nowhere discloses, describes, or refers to an embodiment that does not search for and identify the starting position of an input block that is similar to the output stream. Accordingly, a person of ordinary skill in the art reading the '769 Asserted Claims would understand the disputed terms to mean searching for and identifying the starting position of an input block that is similar to the output stream.

57. Furthermore, a person of ordinary skill in the art would understand that <u>synchronized</u> algorithms – including the prior art "<u>Synchronized</u> Overlap-Add" (SOLA) algorithm that the '769 purports to improve and the "<u>Synchronized</u> Overlap-Add, Fixed Synthesis" algorithm (SOLAFS) that is the purported invention of the '769 patent – require searching for and identifying similar blocks of signal representations. Without such similarity searching/identifying, the claims would pertain to a non-synchronized variety of Overlap-Add algorithms over which SOLA and SOLAFS sought to improve. Thus, a person of ordinary skill in the art reading the disputed terms in the context of the Asserted Claims would understand that the disputed terms must mean "searching for and identifying the starting position of an input block of [W] signal representations that is similar to the output stream."

58. The '769 inventors used the phrase "<u>Synchronized</u> Overlap-Add, Fixed Synthesis method ('SOLAFS')" to describe the algorithm of their purported invention. The inventors describe this algorithm as an improvement over the <u>Synchronized</u> Overlap-Add method ("SOLA") of the prior art:

> The inventive method is an improvement of the prior SOLA method discussed in the Background of the Invention, which inventive method is referred to as the Synchronized Overlap-Add, Fixed Synthesis method ("SOLAFS").

('769 Patent at 7:3-7; *see also id.* at 4:50-61 (SUMMARY OF THE INVENTION . . . The inventive method is an improvement on the SOLA method described in the Background of the Invention and is referred to here as a Synchronized Overlap-Add, Fixed Synthesis time domain processing method ('SOLAFS').").)

59. A person of ordinary skill in the art would unambiguously recognize that because the time-scale modification algorithm of the '769 patent is a "**Synchronized** Overlap-Add, Fixed Synthesis" method, the disputed terms mean "searching for and identifying the starting position of an input block of [W] signal representations that is similar to the output stream." Without such "searching for and identifying" of a "similar" input block, the Asserted Claims would merely pertain to the non-synchronized Overlap-Add methods of the prior art, without the ***synchronization*** required by the purported '769 invention.

60. The '769 patent's "BACKGROUND OF THE INVENTION" recognizes the evolution of the relevant art to include "synchronization." For example, the '769 patent describes prior art from the 1940s that achieved time-scale modification by splicing audio tape. Because these methods did not involve *synchronization* or *overlap-adding*, they resulted in "discontinuities," "clicks," and "pops" during playback:

> Time-domain methods operate by inserting or deleting segments of a speech signal. One of the original time-domain methods of TSM was proposed in the 1940s and entailed splicing, i.e., abutting, different regions of a signal at a fixed rate to compress or expand tape recordings. ***This method results in discontinuities in transitions between inserted or deleted segments and such discontinuities lead to bothersome clicks and pops in the resulting time-scale modified signal.***

(*Id.* at 2:38-46 (emphasis added).)

61. A person of ordinary skill in the art would know and understand – and the '769 patent explains – that the field of the invention developed *synchronization* and *overlap-adding* as methods of "minimiz[ing] the effects of inter-segment transitions":

> Several attempts have been made in the art to ***minimize the effects of inter-segment transitions*** in a time-scale modified signal by improving the splicing method or by windowing adjacent segments. In general, these methods improve quality at the expense of increasing complexity. One such method of time-domain TSM, i.e., "Time-Domain Harmonic Scaling" ("TDHS"), is disclosed in an article entitled "Time-Domain Algorithms for Harmonic Bandwidth Reduction and Time Scaling of Speech Signals" by D. Malah, IEEE Transactions on ASSP, Vol. ASSP-27, April, 1979, pp. 121-133. This article discloses a TDHS algorithm which improves on the original method of splicing by ***synchronizing splice points to a local pitch period and by using overlap-add techniques*** to fade smoothly between the splices. In particular, the TDHS algorithm operates by ***determining the location of each pitch period in the input signal to be modified and then by segmenting the signal around these pitch periods to achieve the desired modification***.

(*Id.* at 2:47-66 (emphasis added).)

62. The "BACKGROUND OF THE INVENTION" section of the '769 patent also describes the "**Synchronized** Overlap-Add method" of Roucos and Wilgus, "High Quality Time-Scale Modification for Speech," *Acoustics, Speech, and Signal Processing, IEEE International Conference on ICASSP*, Vol. 10 (1985) at abstract, 495-496. The Roucos and Wilgus paper (the SOLA paper) is the most significant prior art reference discussed in the '769 patent, because it is the algorithm that the '769 patent seeks to "improve." (*Id.* at 4:57-61.)

63. The Roucos and Wilgus reference describes a "***synchronized*** overlap-and-add algorithm (SOLA)" for time-scale modification that "modifie[s]" a more basic "overlap-and-add (OLA)" method:

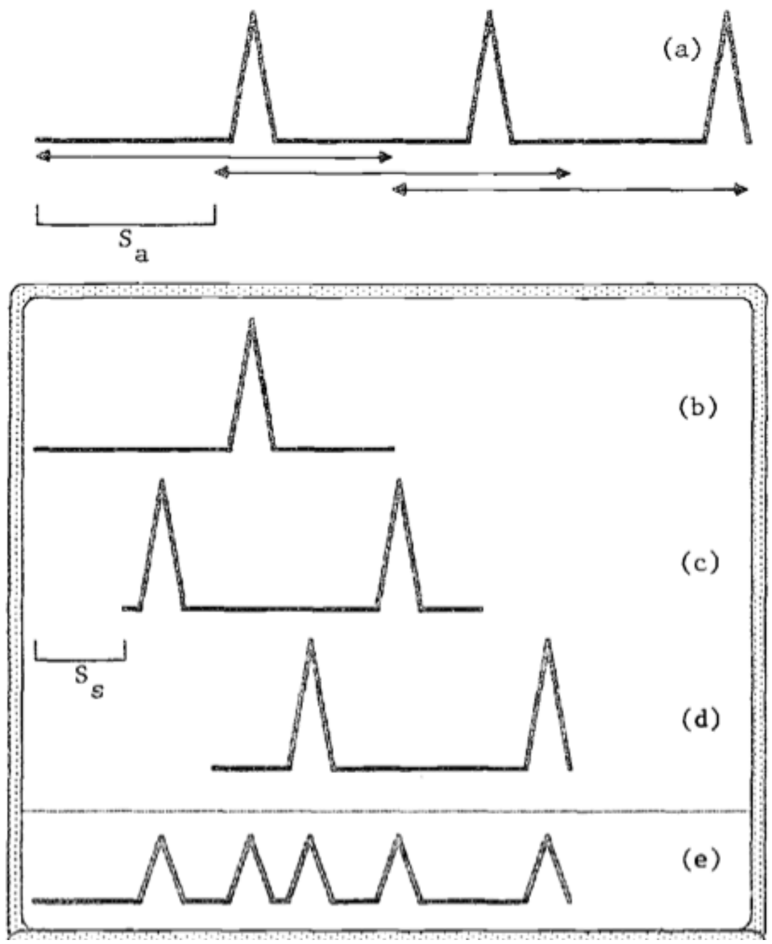
> The algorithm operates in the time domain using a modified overlap-and-add (OLA) procedure on the waveform.

(Roucos and Wilgus, "High Quality Time-Scale Modification for Speech," *Acoustics, Speech, and Signal Processing, IEEE International Conference on ICASSP*, Vol. 10 (1985) ("Roucos and Wilgus 1985") at abstract.)

64. The Roucos and Wilgus reference provides a detailed description of the differences between a ***synchronized*** overlap-add method and a non-synchronized overlap-add method that would be understood by a person of ordinary skill in the art reading the asserted claims of the '769 patent. For example, the Roucos and Wilgus reference explains the shortcomings of a non-synchronized OLA algorithm:

> The first step of the LSEE-MSTFTM algorithm will not change the STFTs, since this first magnitude normalization step consists only of multiplication by unity. The LSEE step will consist of an OLA of windows as shown in Fig. 2. To illustrate the point, we show a pulse train that corresponds to a periodic region of the original waveform. We also show three successive windows (Figs. 2-b, 2-c, and 2-d), collected at intervals of S_a samples, that have been aligned at S_s samples in preparation for the OLA step. The result given by averaging these windows is shown in Fig. 2-e. ***The presence of extraneous pulses explains the tendency of the LSEE-MSTFTM algorithm to sound reverberant after only a small number of iterations***.

(*Id.* at 495 (emphasis added).)



(*Id.* at Fig. 2 (partial).) Note that in Fig. 2, because the OLA method is not synchronized, performing the overlap-add step results in an increase in the number of signal peaks from three to five.

65. Having identified the limitations of a non-synchronized OLA method, the Roucos and Wilgus reference explains that synchronization requires "time-align[ing] the successive windows with respect to signal similarity (magnitude and phase) before the OLA step by maximizing the time-domain cross-correlation between successive windows." (*Id.* at 495.) The Roucos and Wilgus reference further explains that its **synchronized** method overcomes these limitations of a pure OLA method by aligning the overlap-added windows based on a measure of signal similarity:

> In the new TSM algorithm, we propose to time-align the successive windows with respect to signal similarity (magnitude and phase) before the OLA step by maximizing the time-domain crosscorrelation between successive windows. This new initial estimate is given by:

$$x^{o}(n) = \frac{\sum_{m=-\infty}^{\infty} w^{2}(mS_{s}-n)\, y[n-m(S_{s}-S_{a})-k(m)]}{\sum_{m=-\infty}^{\infty} w^{2}(mS_{s}-n)} \quad (6)$$

> ***If k(m)=0, this equation is the same as Step 2 (Eq. 3)*** of the LSEE described above, and we will obtain the same result as shown in Fig. 2-e. However, ***if k(m) is chosen to maximize a crosscorrelation function, we obtain the synchronized overlap-and-add algorithm (SOLA). The choice of k(m) is that value of k that maximizes the normalized cross-correlation*** between the m[th] window of the waveform and the rate-modified signal computed up to the m-1[st] window. Figs. 2-f, 2-g, and 2-h show the successive windows, each shifted by the appropriate k(m) to maximize the crosscorrelation. ***Fig. 2-i shows the average waveform that does not exhibit the extraneous pitch pulses appearing in the signal estimate of Fig. 2-e***.

(*Id.* at 495-6.)

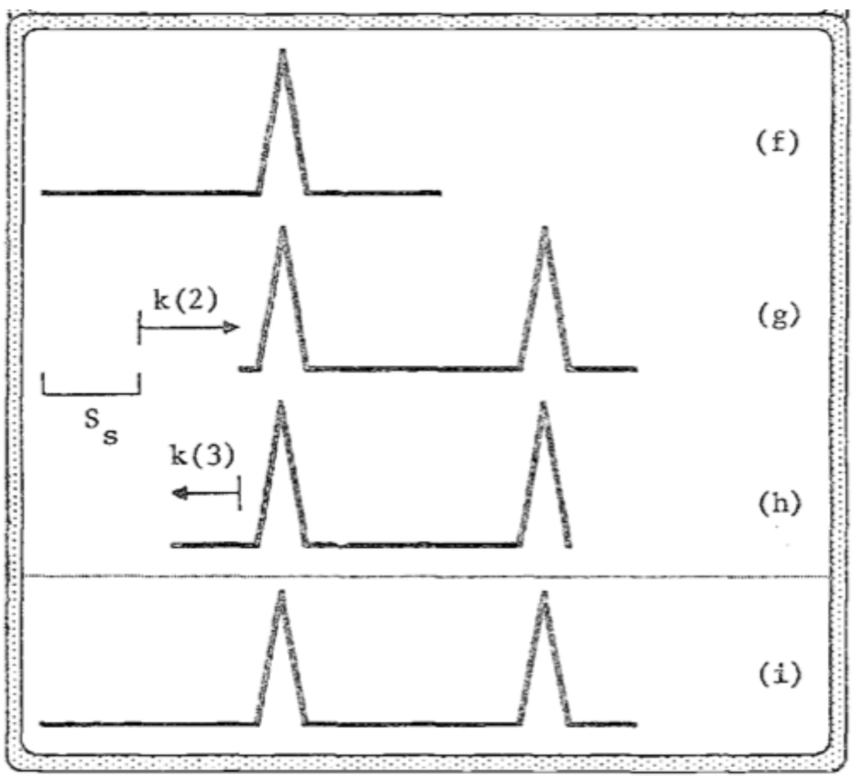


FIGURE 2. Illustrations of OLA and SOLA for TSM

(*Id.* at Fig. 2 (partial).) Notably, Roucos and Wilgus here explain that "[i]f k(m) = 0" – meaning that there is no searching for and identifying of a block of signal representations

based on similarity – the algorithm performed is the non-synchronized OLA method. By contrast, "if k(m) is chosen" based on similarity (in this case, the "maximiz[ation] of a crosscorrelation function"), the resulting signal does not exhibit the extraneous signal peaks of the OLA method, but instead the synchronized overlap-add step results in a decrease in the number of signal peaks from three to two (which preserves the periodicity of the signal). For periodic signals, the overlap-add becomes *synchronized* by the similarity measure to involve an overlap of the signal with itself displaced by an integer number of periods.

66. In short, the Roucos and Wilgus reference identifies that the key distinction between the OLA and SOLA methods is the presence of k(m), which governs whether or not windows of the signal are shifted in order to synchronize the audio signal. Where k(m) is equal to 0, simple fixed-rate overlap-add (OLA) is achieved without synchronization, and the signal of Fig. 2(e) with its "extraneous pulses" causative of reverberation is the result. By contrast, where k(m) is chosen such that the similarity between blocks of overlapped audio is optimized, the signal of Fig. 2(i) is achieved with superior fidelity to the original signal.

67. As explained above, a person of ordinary skill in the art reading the '769 Asserted Claims would understand this distinction between a non-synchronized overlap-add method and a synchronized overlap-add method as described in the Roucos and Wilgus reference. Additionally, the '769 patent also expressly sets forth the "synchronization" aspects of both the prior art Roucos and Wilgus SOLA method and the purportedly inventive SOLAFS method.

68. As explained by the '769 patent, the SOLA and SOLAFS methods are characterized by "analysis" of the input signal followed by "synthesis" of the time-compressed output signal. (*See, e.g.* '769 patent at 3:48-50 ("The SOLA method modifies the time-scale of an input signal in two steps which are referred to as analysis and synthesis, respectively.").) With respect to the SOLA method, the '769 patent explains that the "analysis" phase is "fixed," while the "synthesis" phase is "dynamic." The '769 patent further makes clear that the "dynamic" character of the SOLA synthesis phase relates to the "synchronization" that differentiates the SOLA method from the non-synchronized OLA method that Roucos and Wilgus sought to improve:

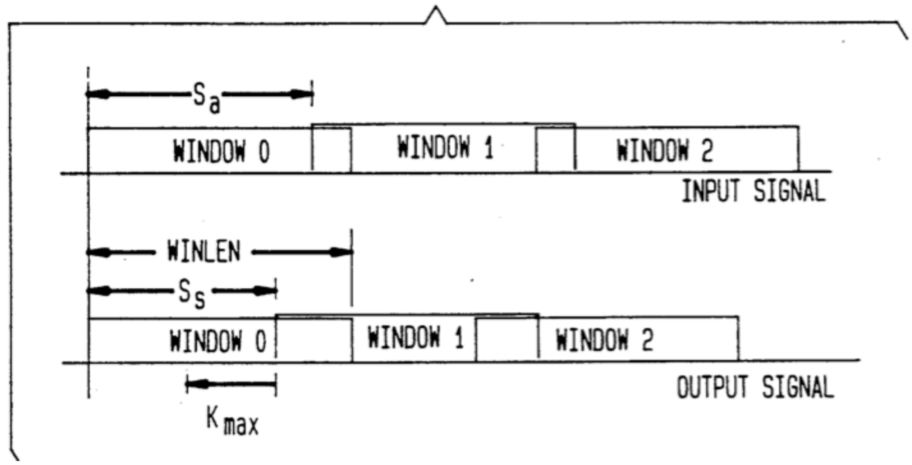
> In accordance with the SOLA method, an input signal is sampled and the samples are segmented at a fixed analysis rate into frames, referred to as windows, and the windows are shifted in time to maintain a predetermined average time-compression or expansion. ***The windows are then overlap-added at a dynamic synthesis rate to provide an output.*** In accordance with this method, the input signal is windowed using a fixed, inter-frame shift interval and the ***output signal is reconstructed using dynamic, inter-frame shift intervals. The inter-frame shift interval used during reconstruction is allowed to vary so that a shift which maximizes the cross-correlation of a current window with previous windows is used. Hence, this method results in a region of overlap which is dynamic between windows and which requires evaluation of a cross-correlation***

> ***with a variable number of points.*** As a result, this method allows one to change the relative overlap between windows which, in turn, modifies the time-scale of the input signal ***without significantly affecting the periods in the signal***.

(*Id.* at 3:12-32 (emphasis added).)

69. The ’769 patent illustrates the SOLA prior art algorithm in connection with Fig. 1. In particular, the ’769 patent illustrates explains that during synthesis, each window “may be shifted for purposes of aligning it with previous windows”:

FIG. 1
(PRIOR ART)



> The SOLA method may be understood in light of the following description which should be read in conjunction with FIG. 1. First, with reference to FIG. 1, there are four parameters which are used in the SOLA method: (a) window length W is the duration of windowed segments of the input signal--this parameter is the same for the input and output buffers and represents the smallest unit of the input signal, for example, speech, that is manipulated by the method; (b) analysis shift S_a is the interframe interval between successive windows along the input signal; (c) synthesis shift S_s is the interframe interval between successive windows along the unshifted output signal; and (d) ***shift search interval K_{max} is the duration of the interval over which a window may be shifted for purposes of aligning it with previous windows***.

(*Id.* at FIG. 1; 3:33-48 (emphasis added).)

70. The ’769 patent further provides a mathematical description of the dynamic synthesis that is required by the Synchronized Overlap-Add Method of the prior art:

> The synthesis shift which is actually used for the m^{th} window $x_m[n]$, i.e., $x_m[n]=x[mS_a+n]$ for n=0, . . . , W-1, is adjusted by an amount k_m which is less than or equal to K_{max} in order to maximize a similarity measure of

> data in the overlapping regions before the overlap-add step is carried out. As a result, in accordance with the SOLA method, the output y[i], where i is a sample index and y[i] is the value of the i^{th} sample, is formed recursively by:
>
> $$y[mS_s+k_m+n] \leftarrow b_m[n]y[mS_s+k_m+n]+(1-b_m[n])x_m[n] \text{ for } n=0, \ldots, W^m{}_{OV}-1 \quad (2)$$
>
> and
>
> $$y[mS_s+k_m+n] \leftarrow x_m[n] \text{ for } n=W^m{}_{OV}, \ldots, W-1 \quad (3)$$
>
> where: $W^m{}_{OV}$ is the number of overlap points for the m^{th} window and $W^m{}_{OV} = k_{m-1} - k_m + W - S_s$. ***Further, shift k_m is selected to maximize a similarity measure, for example, the cross-correlation or average magnitude difference, in the overlap region between the current output y and the m^{th} window x_m.*** Still further, b_m [n] is a fading factor between 0 and 1, for example, an averaging or a linear fade, which is chosen to minimize audible splicing artifacts.

(*Id.* at 4:9-35 (emphasis added).) As the emphasized language indicates, the presence of k_m in this equation reflects the SOLA method's requirement that the output signal is dynamically synthesized (synchronized).

71. Likewise, as set forth above, the purportedly inventive SOLAFS algorithm requires "searching for and identifying the starting position of an input block of [W] signal representations that is similar to the output stream."

2. The '769 Prosecution History

72. As set forth above in Section V, the '769 Prosecution History is brief, as the Examiner allowed the applied-for claims without any rejections or amendments.

73. In a short statement of the Examiner's Reasons For Allowance, the Examiner focused on this "determining an input block for use in overlapping (or appending)" limitation that contains this disputed term:

74. On August 20, 1992, in response to the applicant's original patent application, the PTO mailed a Notice of Allowability allowing claims 1-20. These claims issued as claims 1-20 of the '769 Patent. In this action, the examiner stated:

> EXAMINER REASONS FOR ALLOWANCE; the claimed: 'Time-scale modifying a signal . . . determining an input block for use in overlapping (or appending) . . .' further limited by the other elements of the claim, is not suggested by the prior art.

('769 File History, August. 20, 1992 Notice Of Allowability.)

75. As discussed in the previous section, the '769 patent repeatedly describes a dynamic analysis step as the inventive aspect of the SOLAFS algorithm. While the '769 patent explains that "[i]n accordance with the SOLA method, an input signal is sampled and the samples are segmented at a ***fixed*** analysis rate into frames" ('769 patent at 3:12-14 (emphasis added)), the purportedly inventive '769 patent "***search[es]*** for segments of the input signal near the target starting position mS_a which are similar to the portion of the output signal that will overlap when constructing the output signal." (*Id.* at 5:13-17.) The Examiner's focus on this limitation in his Notice Of Allowability is consistent with my opinion that the apparent disputed terms mean searching for and identifying the starting position of an input block that is similar to the output stream.

3. Extrinsic Evidence

76. The extrinsic evidence further confirms that these disputed terms mean "searching for and identifying the starting position of an input block of [W] signal representations that is similar to the output stream."

77. For example, in a July 1991 publication by the '769 inventors titled "The SOLAFS Time-Scale Modification Algorithm," Hejna and Musicus recognize this meaning for the disputed term as the "key idea" of the SOLAFS algorithm of the '769 patent:

> In this paper we introduce a new variation of SOLA, which we call *Synchronized Overlap-Add, Fixed Synthesis* SOLAFS). ***The key idea is that windows are overlap-added into the output at a fixed synthesis rate, while the starting positions of the windows $x_m[n]$ are adjusted during analysis in order to maximize the similarity between the output and the new window in the overlap region***. Thus during *analysis*, the windows are chosen as:
>
> $$x_m[n] = \begin{cases} x[mS_A + k_m + n] & \text{for } n = 0, \ldots, W-1 \\ 0 & \text{otherwise} \end{cases} \quad (1)$$
>
> and the output is recursively formed by:
>
> $$y[mS_s + n] \leftarrow \begin{cases} \beta[n]y[mS_S + n] + (1-\beta[n])x_m[n] & \text{for } n = 0, \ldots, W_{OV}-1 \\ x_m[n] & \text{for } n = W_{OV}, \ldots, W-1 \end{cases} \quad (2)$$
>
> where $W_{OV} = W - S_s$ is the number of points in the overlap region. ***Note that the shift km now affects where the window starts in the input stream. The optimal shift is determined by maximizing a similarity measure between the overlapping points in x and y.***"

(Hejna and Musicus, "The SOLAFS Time-Scale Modification Algorithm," *BBN*, July 1991, at 2-3 (bold-italic emphasis added); *See also id.* at 6 ("Careful attention should also

be paid to the buffering of data. SOLAFS requires a W_{OV} length output buffer to hold the last points of the output, $y[mS_S], \ldots, y[mS_S + W_{OV} - 1]$, and a $W+K_{max}$: length input buffer to hold the input data points that might be used in the next analysis window, $x[mS_A], \ldots, x\,[mS_A + W + K_{max} - 1]$.”))

78. My opinion that a person of ordinary skill in the art would understand the disputed terms to mean “searching for and identifying the starting position of an input block of [W] signal representations that is similar to the output stream” is further confirmed by Hejna, Donald J., “Real-Time Time-Scale Modification of Speech via the Synchronized Overlap-Add Algorithm” (Thesis), Massachusetts Institute of Technology (“Hejna Thesis”). Notably, ’769 inventors Hejna and Musicus are the author and “Thesis Supervisor,” respectively, of this document.

79. Consistent with my analysis above, the Hejna Thesis recognizes that the distinction between a pure Overlap-Add Algorithm (OLA) and a Synchronized Overlap-Add Algorithm (SOLA) is whether the algorithm includes the searching for and identification of signal representations based on similarity criteria. For example, the Hejna Thesis explains the limitation of the non-synchronized overlap-add algorithms that were prior art to the SOLA algorithm on which the ’769 invention purportedly improves:

> Roucos and Wilgus realized that the overlap-add step of the Griffin-Lim algorithm was working against the desired result. In greatly simplified terms the problem can be summarized as follows: ***If windowed frames of an input signal are taken using an interframe ‘analysis’ shift of Sa sample points, then simply overlap-added with a different interframe ‘synthesis’ shift of Ss sample points, periods in the overlapping portions of the signal will interfere when $S_s \neq S_a$. The interference may be constructive or destructive, and affects various frequencies differently.***
>
> The effect of a simple overlap-add with differing analysis and synthesis shifts will now be examined. Figure 2-3 illustrates a time signal consisting of evenly spaced impulses. This signal is windowed using a fixed interframe interval, S_a, as shown in the diagram labeled *analysis windowing of input signal* of Figure 2-3. ***The windows of input taken with a fixed interframe interval, every S_a points, are then added with S_s points between them to decrease the time-scale of the signal ($S_a < S_a$). Note the spacing between impulses has changed the region of overlap! The overlap-add does not preserve the periodicity of the original signal in the regions of overlap.*** The signal obtained differs greatly from the original input signal when $S_s \neq S_a$. The signal that results has the desired time-scale, but no longer represents the original input. ***This modified signal is referred to as the rate-modified-unshifted-signal in Figures 2-2 and 2-3.***

(Hejna Thesis at 9 (bold-italic emphasis added).)

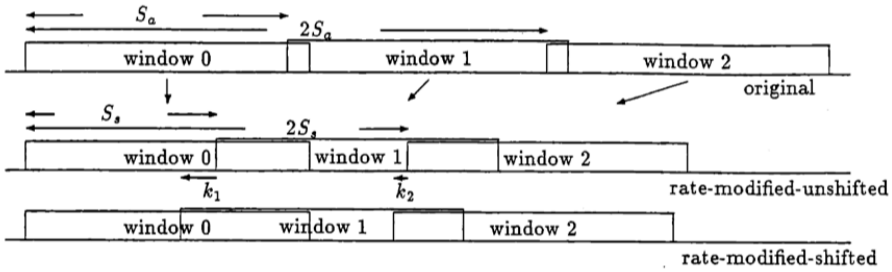


Figure 2-2: Overview of time-scale compression.

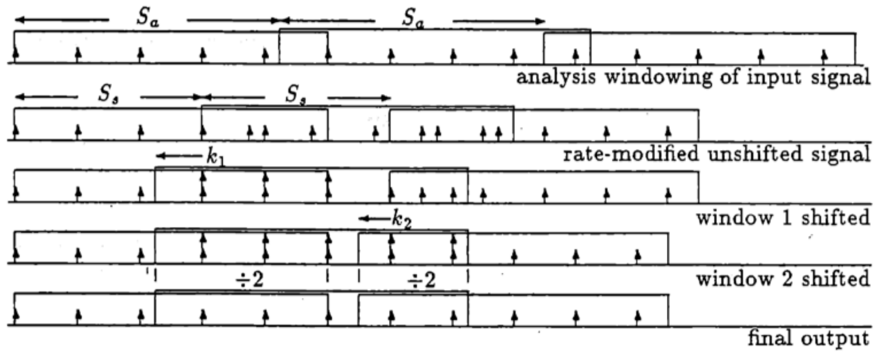


Figure 2-3: Detail of Synchronized Overlap-Add for time-scale compression.

(*Id.* at Figs. 2-2 and 2-3; *see also id.* at 9-10 (explaining the role of phase in creating discontinuities with respect to a non-synchronized Overlap-Add Algorithm).)

80. The Hejna Thesis further distinguishes the prior art OLA algorithm from SOLA based on SOLA's alignment of windows based on similarity:

> Roucos and Wilgus proposed time-aligning windows on the basis of signal similarity before adding them. This is accomplished by maximizing the crosscorrelation of overlapping windows. The crosscorrelation aligns the windows before the overlap-add step, preserving the magnitude and phase (i.e., the local period). ***They dubbed this process "Synchronized Overlap-Add" or "SOLA" to distinguish it from the Griffin-Lim "Overlap-Add" which performs no shifting.***

(*Id.* at 10 (emphasis added).)

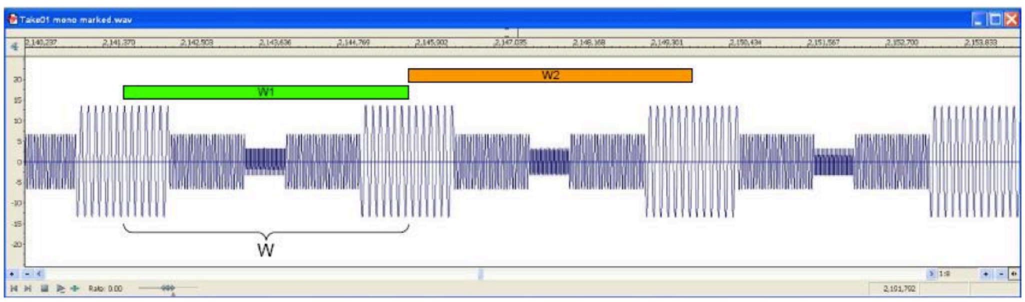
81. Thus, the Hejna Thesis supports my opinion that a person of ordinary skill in the art reading the '769 Asserted Claims would have understood that the purportedly inventive "Synchronized Overlap-Add Algorithm, Fixed Synthesis" of the '769 patent requires searching for and identifying the starting position of the input block of signal representations that is similar to the output stream in the overlapping interval.

4. EPL's Proposed Construction

82. EPL's proposal that these terms would have a plain and ordinary meaning to a person of ordinary skill in the art is incorrect.

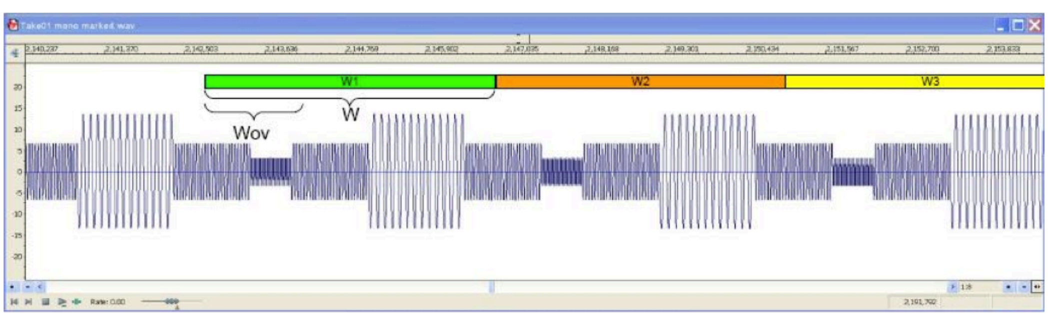
83. I have reviewed Plaintiff EPL Holding, LLC's Amended Disclosure Of Asserted Claims And Infringement Contentions dated March 22, 2013 ("EPL Infringement Contentions"). Although EPL contends that the disputed terms should be accorded their plain and ordinary meaning, Exhibit A indicates that EPL interprets the disputed terms to include merely segmenting the input signal at a fixed analysis rate:

> The '769 Patent Accused Product determines an input block of W signal representations from the input stream for use in overlapping signal representations from the input block with signal representations in the output stream.
>
> Referring the test example, an input block of W signal representations from the input stream is shown here:



(EPL Infringement Contentions, Ex. A at 2, 9-10.)

> The '769 Patent Accused Product determines an input block of W signal representations from the input stream for use in overlapping signal representations from the input block with signal representations in the output stream.
>
> Referring to the test example, an input block of W signal representations from the input stream is shown here:



(*Id.* at 7.)

84. Such an interpretation of the disputed terms to include merely segmenting the input signal at a fixed analysis rate would fail to distinguish the Asserted Claims from the

prior art under the '769 patent's own characterization of the prior art. As recognized by the '769 patent:

> ***In accordance with the SOLA method, an input signal is sampled and the samples are segmented at a fixed analysis rate into frames, referred to as windows***, and the windows are shifted in time to maintain a predetermined average time-compression or expansion. The windows are then overlap-added at a dynamic synthesis rate to provide an output. ***In accordance with this method, the input signal is windowed using a fixed, inter-frame shift interval*** and the output signal is reconstructed using dynamic, inter-frame shift intervals. The inter-frame shift interval used during reconstruction is allowed to vary so that a shift which maximizes the cross-correlation of a current window with previous windows is used.

(*Id.* at 3:12-24 (emphasis added).) Likewise, the prior art non-synchronized Overlap-Add method includes merely segmenting the input signal at a fixed analysis rate (*see* above), and thus EPL's interpretation of the disputed terms would also fail to distinguish the Asserted Claims from such OLA methods.

B. "W"

Claim Nos.	Proposed Term	Apple's Proposed Construction	EPL's Proposed Construction
1, 2, 10, 11	W	"a parameter that fixes the duration of the windowed segments of the input signal that represents the smallest unit that the time-scale modification method manipulates"	"A parameter that represents the duration of the windowed segments of the input signal"

85. A person of ordinary skill in the art reading the term "W" in the context of the asserted '769 claims would understand this term to mean "a parameter that fixes the duration of the windowed segments of the input signal that represents the smallest unit that the time-scale modification method manipulates."

1. The '769 Patent Claims, Specification, and Figures

86. The plain language of claims 1 and 10 indicates that the term "W" is a parameter that establishes the quantity of signal representations that form "an input block . . . from the input stream for use in overlapping signal representations from the input block with signal representations in the output stream." ('769 patent at claims 1, 10, and 19.) This requirement recited in the claims is consistent with a construction of "W" as "a parameter that fixes the duration of the windowed segments of the input signal that represents the smallest unit that the time-scale modification method manipulates."

87. A person of ordinary skill in the art reading the '769 patent claims would recognize that W is a parameter defined by and used in the purportedly inventive SOLAFS algorithm set forth in the patent specification. In the context of mathematical algorithms and formulae, it is well-understood to persons of ordinary skill in the art that such parameters have no meaning independent of how they are defined for the purposes of the algorithm in which they are used. Thus, a person of ordinary skill in the art would refer to the definition of "W" provided by the '769 specification.

88. The '769 patent specification establishes that "W" means "a parameter that fixes the duration of the windowed segments of the input signal that represents the smallest unit that the time-scale modification method manipulates." For example, the '769 specification states:

> The inventive method is an improvement of the prior SOLA method discussed in the Background of the Invention, which inventive method is referred to as the Synchronized Overlap-Add, Fixed Synthesis method ("SOLAFS"). With reference to FIGS. 1 and 2, ***there are four parameters which are used in the inventive SOLAFS method***: (a) ***window length W is the duration of windowed segments of the input signal--this parameter is the same for input and output buffers and represents the smallest unit of the input signal, for example, speech, that is manipulated by the method*** . . .

(*Id.* at 7:3-13 (emphasis added).)

FIG. 1
(PRIOR ART)

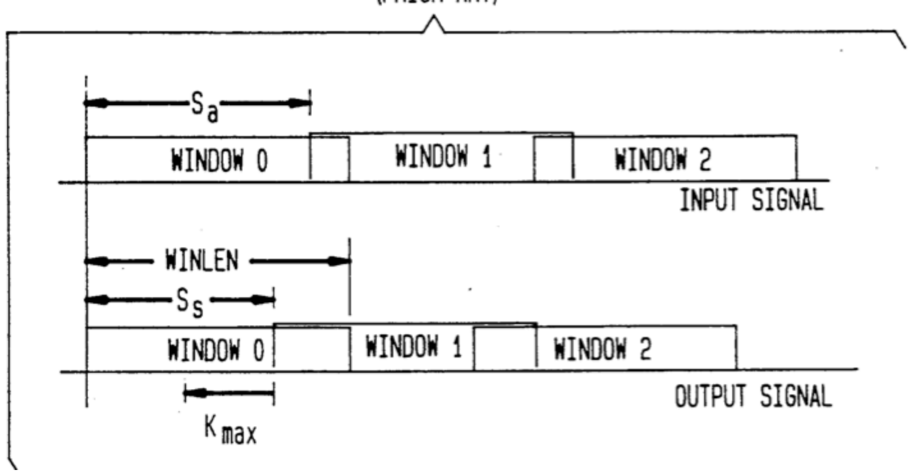
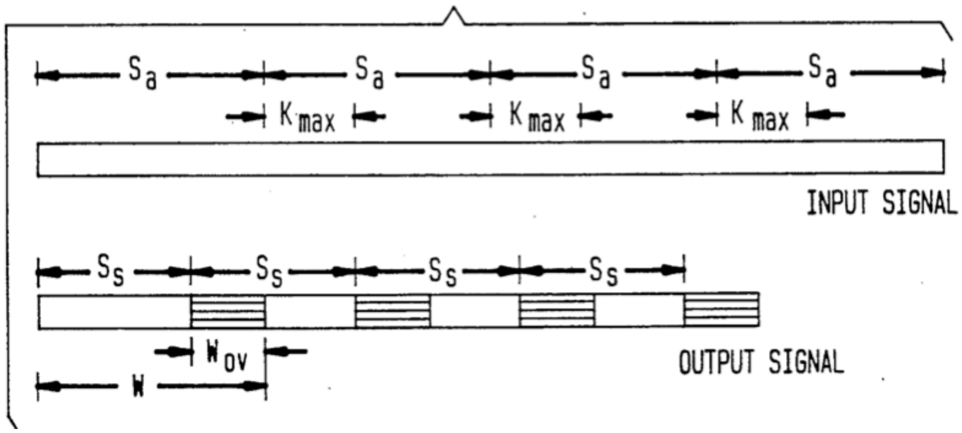


FIG. 2



(*Id.* at FIGS. 1 and 2.)

89. Likewise, in connection with the examples set forth in Figs. 5-7, the '769 specification explains that "W" means "a parameter that fixes the duration of the windowed segments of the input signal that represents the smallest unit that the time-scale modification method manipulates." For example, the '769 specification states:

> FIGS. 5-7 show a flowchart of one embodiment of the inventive SOLAFS method. The following is nomenclature which is used in the following flowchart: ***(a) W is the window length and represents the smallest block or unit of a signal that is manipulated by the inventive method*** . . . .

(*Id.* at 12:42-47.) Significantly, the '769 patent discloses no embodiments where "W" does not mean "a parameter that fixes the duration of the windowed segments of the input signal that represents the smallest unit that the time-scale modification method manipulates." (*See, e.g. id.* at 8:67 – 10:4; 10:66 – 11:15; FIGS. 4 and 8.)

90. The '769 patent's equations for the SOLAFS algorithm also demonstrate that "W" means "a parameter that fixes the duration of the windowed segments of the input signal that represents the smallest unit that the time-scale modification method manipulates," as they indicate that the SOLAFS algorithm operates over W signal representations in each of its analysis and synthesis steps:

> In the following x[i] represents the ith sample in the input digital stream representative of an input signal. In accordance with the inventive method, analysis windows are chosen as follows:

$$x_m[n] = \begin{cases} x_m[mS_a + k_m + n] & \text{for } n = 0, \ldots, W-1 \\ 0 & \text{otherwise} \end{cases} \quad (4)$$

> where: m is a window index, i.e., it refers to the m^{th} window; n is a sample index in an input buffer for the input signal, which buffer is W samples long; k_m is the number of samples of shift for the m^{th} window; and $x_m[n]$ represents the nth sample in the m^{th} analysis window.
>
> The analysis windows are then used to form the output signal y[i] recursively in accordance with the following:
>
> $$y[mS_s+n] \leftarrow b[n]y[mS_s+n]+(1-b[n])x_m[n] \text{ for } n=0, \ldots, W_{OV}-1 \quad (5)$$
>
> and
>
> $$y[mS_s+n] \leftarrow x_m[n] \text{ for } n=W_{OV}, \ldots, W-1 \quad (6)$$
>
> where: W_{OV} =W-S_s is the number of points in the overlap region and b[n] is an overlap-add weighting function which is referred to as a fading factor--an averaging function, a linear fade function, and so forth.

(*Id.* at 7:48 – 8:9.)

91. The '769 patent also explains that manipulating windows of such fixed length "W" are an important advantage of its purported invention:

> Since the inventive method uses ***fixed segment lengths*** which are independent of local pitch, the inventive SOLAFS method advantageously operates equally well on speech or non-speech signals.

(*Id.* at 5:65-68 (emphasis added); *see also id.* at 3:54-56 (In the SOLA method, "[e]ach input window has a fixed length W . . .").)

2. EPL's Proposed Construction

92. EPL's proposed construction is incomplete and incorrect.

93. First, EPL's proposed construction ignores that the specification defines that "W" "represents the smallest unit that the time-scale modification method manipulates." *See above*.

94. Second, EPL's proposed construction is incorrect because it does not specify that "W" is a parameter that "fixes" the duration of the windowed segments of the input signal. Thus, EPL's construction would apparently permit "W" to act as a variable. This aspect of EPL's construction is incorrect for at least three reasons.

95. Construing "W" as a variable is inconsistent with the specification, which makes clear that "W" is a fixed quantity. In particular, "W" "represents the smallest unit that the

time-scale modification method manipulates," and therefore cannot assume multiple values. In other words, there may only be one "smallest" value.

96. The '769 specification further explains that "W" is a fixed quantity. For example:

> Since the inventive method uses ***fixed segment lengths*** which are independent of local pitch, the inventive SOLAFS method advantageously operates equally well on speech or non-speech signals.

(*Id.* at 5:65-68.) The '769 patent also defines W in relation to two other fixed variables (i.e., $W_{ov} = W - S_s$). Because W_{ov} and S_s are each fixed (*see* sections below), as a matter of arithmetic, W must also be fixed.

97. Furthermore, the '769 specification makes clear that "W" is not a variable, but it is a "parameter." For example:

> The inventive method is an improvement of the prior SOLA method discussed in the Background of the Invention, which inventive method is referred to as the Synchronized Overlap-Add, Fixed Synthesis method ("SOLAFS"). With reference to FIGS. 1 and 2, there are four ***parameters*** which are used in the inventive SOLAFS method: (a) window length W is the duration of windowed segments of the input signal--this parameter is the same for input and output buffers and represents the smallest unit of the input signal, for example, speech, that is manipulated by the method . . .

(*Id.* at 7:3-13 (emphasis added).) A person of ordinary skill in the art would understand that, in mathematics, a variable is a value that may change within the scope of a given problem or set of operations. By contrast, a parameter is a non-varying value – or a value that is fixed in the context of use. To illustrate the difference, the quadratic formula contains both variables (x and y) and parameters (a, b, and c): $y = ax^2 + bx + c$. In operation, for a given quadratic function, a, b, and c are fixed, but variables x and y vary. Likewise, in the context of the '769 specification, the SOLAFS algorithm's "parameters" must be fixed in order to establish the conditions for a particular mode of time-scale modification.

C. "W_{OV}"

Claim Nos.	Proposed Term	Apple's Proposed Construction	EPL's Proposed Construction
1, 2, 10, 11	W_{OV}	"a parameter that fixes the number of signal representations to be overlapped as determined by W and the time-scale modification"	"a parameter that represents the number of signal representations to be overlapped as determined by W and the time-scale modification"

98. A person of ordinary skill in the art reading the term "W_{OV}" in the context of the asserted '769 claims would understand this term to mean "a parameter that fixes the number of signal representations to be overlapped as determined by W and the time-scale modification."

1. The '769 Patent Claims, Specification, and Figures

99. The plain language of the Asserted Claims indicates that "W_{ov}" is a parameter that establishes the quantity of "overlapping . . . signal representation from the beginning of the input block with W_{ov} signal representations from the end of the output stream, where W_{ov} is determined by W and the time-scale modification." ('769 patent at claim 1.) These requirements recited in the claims are consistent with a construction of "W_{ov}" as "a parameter that fixes the number of signal representations to be overlapped as determined by W and the time-scale modification."

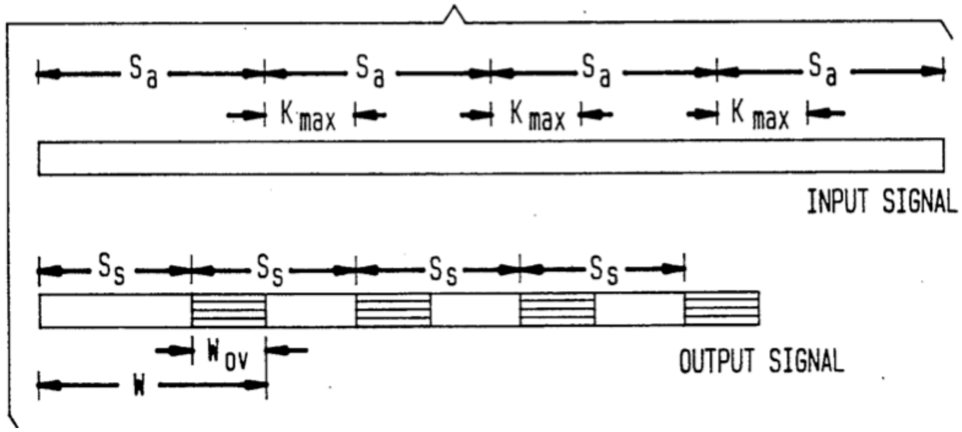
100. As with "W," a person of ordinary skill in the art reading the '769 patent claims would recognize that "W_{ov}" is a parameter defined by and used in the purportedly inventive SOLAFS algorithm set forth in the patent specification. In the context of mathematical algorithms and formulae, it is well-understood to persons of ordinary skill in the art that such parameters have no meaning independent of how they are defined for the purposes of the algorithm in which they are used. Thus, a person of ordinary skill in the art would refer to the definition of "W_{ov}" provided by the '769 specification.

101. The '769 specification establishes that "W_{ov}" means "a parameter that fixes the number of signal representations to be overlapped as determined by W and the time-scale modification." For example, the '769 specification states:

> In essence, ***the first W_{ov} samples in each new window in the input signal, referred to as an analysis window, are overlap-added with the last W_{ov} samples in the output signal, i.e., this is referred to as overlap-adding at a fixed synthesis rate.*** In accordance with the inventive method, the starting point of each analysis window is varied by: ***(a) evaluating a similarity measure such as, for example, the cross-correlation, of the first W_{ov} points in the analysis window with the last W_{ov} points in the output signal, where W_{ov} is a predetermined, fixed number***; (b) then the starting point of the analysis window is shifted by a fixed amount and a new cross-correlation of the first W_{ov} points in the new analysis window with the same last W_{ov} points in the output signal is evaluated; (c) step (b) is performed a predetermined number of times, Kmax, and the new analysis window is chosen to be the one wherein the cross-correlation is maximized. ***Finally, the first W_{ov} samples in the new analysis window are overlap-added with the last W_{ov} samples in the output signal and Ss additional points from the analysis window are appended to the output signal. The term overlap-added refers to a method of combination such as averaging points or performing a weighted average in accordance with a predetermined weighting function***.

(*Id.* at 7:23-47.)

FIG. 2



(*Id.* at FIG. 2.)

102. The ’769 patent’s equations for the SOLAFS algorithm also demonstrate that “W_{ov}” means “a parameter that fixes the number of signal representations to be overlapped as determined by W and the time-scale modification,” as they indicate that the SOLAFS algorithm operates over W_{ov} signal representations from the input stream and the output stream in order to effect overlap-adding (according to weighting function b(n)) and that W_{ov} is determined by W and the time-scale modification:

> In the following x[i] represents the ith sample in the input digital stream representative of an input signal. In accordance with the inventive method, analysis windows are chosen as follows:
>
> $$x_m[n] = \begin{cases} x_m[mS_a + k_m + n] & \text{for } n = 0, \ldots, W-1 \\ 0 & \text{otherwise} \end{cases} \quad (4)$$
>
> where: m is a window index, i.e., it refers to the m^{th} window; n is a sample index in an input buffer for the input signal, which buffer is W samples long; $\mathbf{k_m}$ is the number of samples of shift for the m^{th} window; and $x_{\mathbf{m}}$ [n] represents the nth sample in the m^{th} analysis window.
>
> The analysis windows are then used to form the output signal y[i] recursively in accordance with the following:
>
> $$y[mS_s+n] \leftarrow b[n]y[mS_s+n]+(1-b[n])x_m[n] \text{ for } n=0, \ldots, W_{OV}-1 \quad (5)$$

and

$$y[mS_s+n] \leftarrow x_m[n] \text{ for } n=W_{OV}, \ldots, W-1 \quad (6)$$

where: W_{OV} =W-S_s is the number of points in the overlap region and b[n] is an overlap-add weighting function which is referred to as a fading factor--an averaging function, a linear fade function, and so forth.

(*Id.* at 7:48 – 8:9; *see also id.* at 8:67 – 10:20.)

103. The '769 patent is clear that W_{ov} is fixed for a particular time-scale modification. In fact, it is the purportedly novel "fixed synthesis" aspect of the '769 patent's "Synchronized Overlap-Add, **Fixed** Synthesis time domain processing method ('SOLAFS')" that results in a fixed W_{ov}. For example, the '769 patent states:

> Method for time-scale modification ("TSM") of a signal, for example, a voice signal, wherein starting positions of blocks in an input signal, referred to as analysis windows, are varied and ***an output signal is reconstructed by overlapping analysis windows using fixed window offsets, i.e., the duration of overlap between analysis windows is fixed during reconstruction***. . . . ***[T]he predetermined number of samples from the beginning of the analysis window are averaged with the predetermined number of samples from the end of the previous portion of the output signal*** and the remaining samples in the window are appended to the averaged segment of the previous portion of the output signal.

(*Id.* at abstract (emphasis added).)

> ***[T]he first WOV samples in each new window in the input signal, referred to as an analysis window, are overlap-added with the last WOV samples in the output signal, i.e., this is referred to as overlap-adding at a fixed synthesis rate***. In accordance with the inventive method, the starting point of each analysis window is varied by: (a) ***evaluating a similarity measure such as, for example, the cross-correlation, of the first WOV points in the analysis window with the last WOV points in the output signal, where WOV is a predetermined, fixed number***; (b) then the starting point of the analysis window is shifted by a fixed amount and a new cross-correlation of the first WOV points in the new analysis window with the same last WOV points in the output signal is evaluated; (c) step (b) is performed a predetermined number of times, Kmax, and the new analysis window is chosen to be the one wherein the cross-correlation is maximized. ***Finally, the first WOV samples in the new analysis window are overlap-added with the last WOV samples in the output signal*** and Ss additional points from the analysis window are appended to the output signal. The term overlap-added refers to a method of combination such as

> averaging points or performing a weighted average in accordance with a predetermined weighting function.

(*Id.* at 7:23-47 (emphasis added); *see also id.* at 12:56-57 ("W_{OV} = W – Ss is the fixed number of overlapping points between windows").)

104. Significantly, the '769 further asserts that the lack of a fixed overlap region W_{ov} was a "drawback" in the prior art SOLA method, and that the fixed overlap region W_{ov} is an important advantage of the purportedly inventive SOLAFS method:

> ***The SOLA method has a drawback in that the amount of overlap for the m^{th} window, $W^m{}_{OV}$, between the output and the m^{th} analysis window varies with k_m and this complicates the work required to compute the similarity measure and to fade across the overlap region***. Also, depending on the shifts k_m, more than two windows may overlap in certain regions and this further complicates the fading computation.
>
> ***As a result, there is a need*** in the art for a method for modifying the time-scale of speech, music, or other acoustic material without modifying the pitch, which is robust, and which ***does not require excessive amounts of computation***.

(*Id.* at 3:36-4:48 (emphasis added).)

> In accordance with the present invention, blocks of the input signal, referred to as analysis windows, are taken at an average rate of Sa with each starting position allowed to vary within limits and an output signal is reconstructed using a fixed inter-block offset Ss, i.e., the duration of overlap with the existing signal in each window to be added is fixed. This is done by searching for segments of the input signal near the target starting position mSa which are similar to the portion of the output signal that will overlap when constructing the output signal. A similarity measure is used to evaluate such similarity and, in accordance with the present invention, the similarity measure uses a fixed, predetermined minimum number of samples. ***The fact that the region of overlap is fixed is advantageous because the number of computations which are required to evaluate the similarity measure over the range of shift values are reduced over that required in the prior art SOLA method.*** Several similarity measures are evaluated by shifting the starting point of an analysis window over a predetermined number of samples, i.e., removing samples from the beginning of the analysis window as new samples from the input are appended to the tail of the analysis window, thus using the same, predetermined number of samples in the evaluation. The starting position of the analysis window which provides the maximum similarity in the region of the analysis window which will overlap with the region of the output signal is selected from all starting positions tested. Finally, ***the predetermined number of samples in the region of overlap are combined***

> ***with the predetermined number of samples from the end of the previous portion of the output signal*** and the remaining samples in the window are appended to the combined segment of the previous portion of the output signal.

(*Id.* at 5:7-41 (emphasis added).)

> Still further, ***since the inventive method maintains the extent of superimposition constant throughout each frame and fixes it over the range of reproduction rates, the inventive SOLAFS method advantageously simplifies the computation required when compared to the computation required to carry out the SOLA method***. As a result, the inventive SOLAFS method advantageously provides a robust time-scale modification ("TSM") signal using substantially less computation than SOLA or TDHS and the TSM signal is unaffected by the presence of white noise in the input signal. Further, using a relatively small amount of trial and error, one can determine parameters for use in embodying the inventive method so that the resultant time-scale modified speech contains few audible artifacts and preserves speaker identity.

(*Id.* at 6:5-21 (emphasis added).)

> Finally, note that ***overlap regions occur in the output with a predictable rate, S_s, and have a fixed length, W_{OV}***. This can be seen in FIG. 2 which shows a TSM compressed signal and FIG. 3 which shows a TSM expanded signal. ***Therefore, a fixed-length fading function b[n] can be used, and its values can be precomputed and stored in a lookup table.***

(*Id.* at 8:51-57 (emphasis added).)

FIG. 2

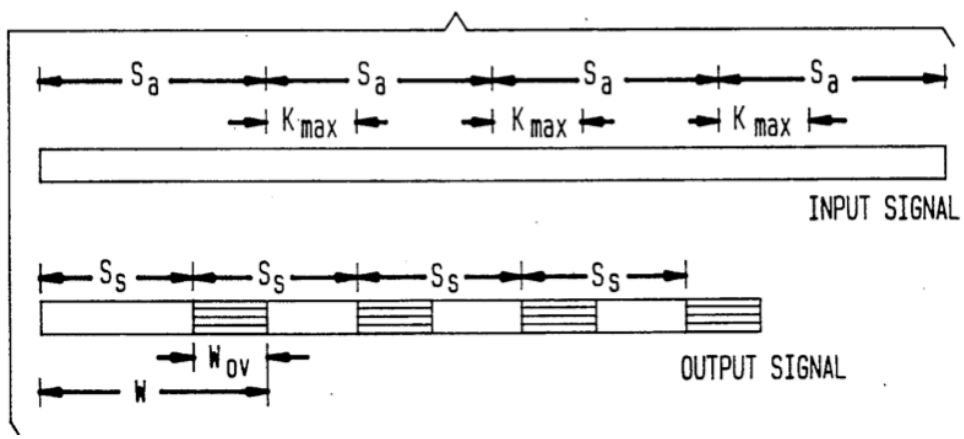
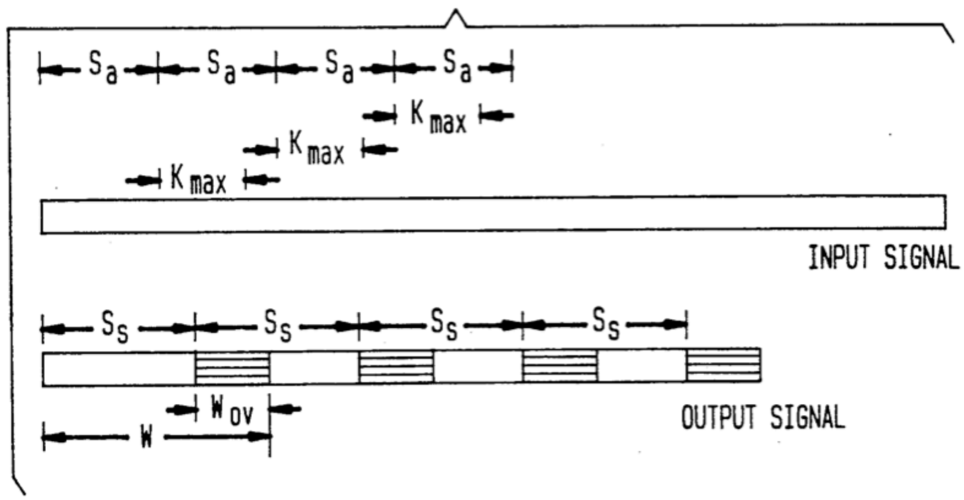


FIG. 3



(*Id.* at FIGS. 2 and 3.)

105. Notably, as shown above, the '769 patent arithmetically defines W_{ov} in relation to W and S_s. Because each of W and S_s is fixed, W_{ov} must also be fixed.

2. EPL's Proposed Construction

106. The latter part of Apple's and EPL's respective proposed constructions is identical. Thus, they differ only in the underlined regions indicated below:

> Apple's Proposed Construction: "a parameter that <u>fixes</u> the number of signal representations to be overlapped as determined by W and the time-scale modification"

> EPL's Proposed Construction: "a parameter that <u>represents</u> the number of signal representations to be overlapped as determined by W and the time-scale modification"

107. EPL's proposed construction is incorrect.

108. First, as described above, the specification makes clear that "W_{ov}" is a fixed quantity. In fact, the specification further makes clear that it is the overlap region of the SOLA method that was "variable," unlike the "fixed" overlap region of the purportedly inventive SOLAFS method:

> [The SOLA] method results in ***a region of overlap which is dynamic between windows*** and which requires evaluation of a cross-correlation with a ***variable*** number of points. As a result, ***this method allows one to change the relative overlap between windows*** which, in turn, modifies the time-scale of the input signal without significantly affecting the periods in the signal.

(*Id.* at 3:25-32 (emphasis added).)

109. Second, as described above at Section VI.B.2, a person of ordinary skill in the art would understand that, because W_{ov} is a parameter of the purportedly inventive SOLAFS algorithm, it must be fixed in order to establish the conditions for a particular mode of time-scale modification.

D. “S_s”

Claim Nos.	Proposed Term	Apple’s Proposed Construction	EPL’s Proposed Construction
2, 11	S_s	“a parameter that fixes the interframe interval between successive windows of length W along the output signal”	“A parameter that represents the interframe interval between successive analysis windows along the output signal”

110. A person of ordinary skill in the art reading the term “S_s” in the context of the asserted ’769 claims would understand this term to mean “a parameter that fixes the interframe interval between successive windows of length W along the output signal.”

1. The ’769 Patent Claims, Specification, and Figures

111. The plain language of the Asserted Claims indicates that the term “S_s” is a parameter that establishes the quantity of “signal representations from the input stream [placed] at the end of the output stream . . . subsequent to the W_{ov} signal representations from the beginning of the input block.” (’769 patent at claims 2, 11.) The requirements recited in the claims are consistent with a construction of “S_s” as “a parameter that fixes the interframe interval between successive windows of length W along the output signal.”

112. As with W and W_{ov}, A person of ordinary skill in the art reading the ’769 patent claims would recognize that S_s is a parameter defined by and used in the purportedly inventive SOLAFS algorithm set forth in the patent specification. In the context of mathematical algorithms and formulae, it is well understood to persons of ordinary skill in the art that such parameters have no meaning independent of how they are defined for the purposes of the algorithm in which they are used. Thus, a person of ordinary skill in the art would refer to the definition of “S_s” provided by the ’769 specification.

113. The ’769 patent specification establishes that “S_s” means “a parameter that fixes the interframe interval between successive windows of length W along the output signal.” For example, the ’769 specification states:

> The inventive method is an improvement of the prior SOLA method discussed in the Background of the Invention, which inventive method is referred to as the Synchronized Overlap-Add, Fixed Synthesis method ("SOLAFS"). With reference to FIGS. 1 and 2, there are four ***parameters***

which are used in the inventive SOLAFS method: (a) window length W is the duration of windowed segments of the input signal--this parameter is the same for input and output buffers and represents the smallest unit of the input signal, for example, speech, that is manipulated by the method; (b) analysis shift S_a is the interframe interval between successive search ranges for analysis windows along the input signal; (c) ***synthesis shift S_s is the interframe interval between successive analysis windows along the output signal***; and (d) shift search interval K_{max} is the duration of the interval over which an analysis window may be shifted for purposes of aligning it with the region of the output signal it will overlap.

FIG. 1
(PRIOR ART)

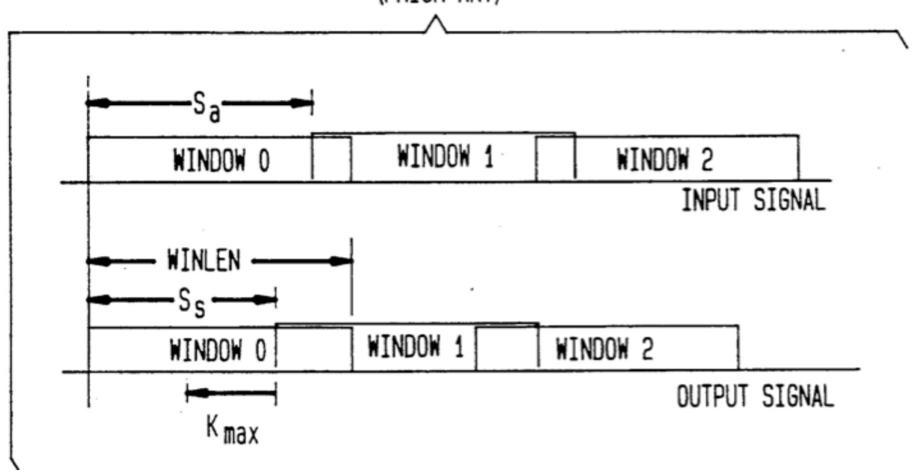
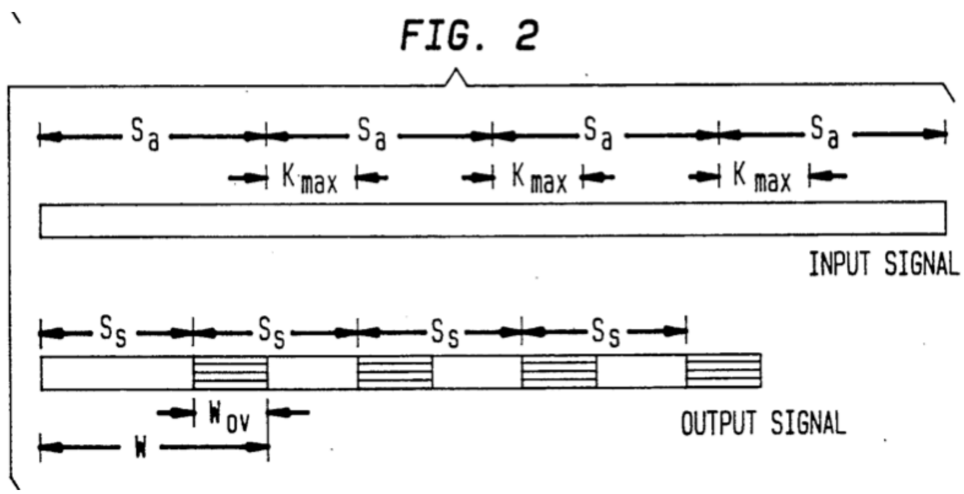


FIG. 2



(*Id.* at FIGS. 1 and 2.)

114. Likewise, in connection with the examples set forth in Figs. 5-7, the '769 specification explains that "Ss" means "a parameter that fixes the interframe interval between successive windows of length W along the output signal":

> FIGS. 5-7 show a flowchart of one embodiment of the inventive SOLAFS method. The following is nomenclature which is used in the following flowchart: (a) W is the window length and represents the smallest block or unit of a signal that is manipulated by the inventive method; (b) Sa is the analysis shift and represents the interframe interval between successive search intervals along the input signal; (c) ***S_s is the synthesis shift and represents the interframe interval between successive windows in the output signal***; (d) k_m is the window shift and represents the number of data samples the mth analysis window is shifted from its target position, mSa, to provide alignment with previous windows; (e) K_{max} is the maximum window shift, i.e., $0 \leq k_m \leq K_{max}$ for all m; (f) $W_{OV} = W - S_s$ is the fixed number of overlapping points between windows; (g) head-- buf is a storage buffer for samples from an input signal buffer, head-- buf has a length of K_{max} +W; and (h) tail-- buf is a storage buffer of length W_{OV}.

(*Id.* at 12:42-60 (emphasis added).)

115. The '769 patent's equations for the SOLAFS algorithm also demonstrate that "a parameter that fixes the interframe interval between successive windows of length W along the output signal," as they indicate that successive windows are spaced S_s signal representations apart during synthesis:

> The analysis windows are then used to form the output signal y[i] recursively in accordance with the following:
>
> $$y[mS_s+n] \leftarrow b[n]y[mS_s+n]+(1-b[n])x_m[n] \text{ for } n=0, \ldots, W_{OV}-1 \quad (5)$$
>
> and
>
> $$y[mS_s+n] \leftarrow x_m[n] \text{ for } n=W_{OV}, \ldots, W-1 \quad (6)$$
>
> where: $W_{OV} = W - S_s$ is the number of points in the overlap region and b[n] is an overlap-add weighting function which is referred to as a fading factor--an averaging function, a linear fade function, and so forth.

(*Id.* at 7:64 – 8:9.)

116. The '769 patent is clear that S_s is fixed for a particular time-scale modification. In fact, it is the purportedly novel "fixed synthesis" aspect of the '769 patent's "Synchronized Overlap-Add, **<u>Fixed</u>** Synthesis time domain processing method ('SOLAFS')" that results in a fixed S_s. For example, the '769 patent states:

> Method for time-scale modification ("TSM") of a signal, for example, a voice signal, wherein starting positions of blocks in an input signal, referred to as analysis windows, are varied and ***an output signal is reconstructed by overlapping analysis windows using fixed window offsets***, i.e., the duration of overlap between analysis windows is fixed during reconstruction.

(*Id.* at abstract (emphasis added).)

> ***In accordance with the present invention***, blocks of the input signal, referred to as analysis windows, are taken at an average rate of Sa with each starting position allowed to vary within limits and ***an output signal is reconstructed using a fixed inter-block offset*** S_s, i.e., the duration of overlap with the existing signal in each window to be added is fixed. This is done by searching for segments of the input signal near the target starting position mS_a which are similar to the portion of the output signal that will overlap when constructing the output signal. A similarity measure is used to evaluate such similarity and, in accordance with the present invention, the similarity measure uses a fixed, predetermined minimum number of samples. ***The fact that the region of overlap is fixed is advantageous because the number of computations which are required to evaluate the similarity measure over the range of shift values are reduced over that required in the prior art SOLA method***.

(*Id.* at 5:7-25 (emphasis added).)

> ***[N]ote that overlap regions occur in the output with a predictable rate,*** S_s***, and have a fixed length,*** W_{OV}. This can be seen in FIG. 2 which shows a TSM compressed signal and FIG. 3 which shows a TSM expanded signal. ***Therefore, a fixed-length fading function b[n] can be used, and its values can be precomputed and stored in a lookup table***.

(*Id.* at 8:51-57 (emphasis added).)

117. Notably, as shown above, the '769 patent arithmetically defines S_s in relation to W and W_{OV}. Because each of W and W_{OV} is fixed, S_s must also be fixed.

2. EPL's Proposed Construction

118. EPL's proposed construction is incorrect.

119. First, as described above, the specification makes clear that "S_s" is a fixed quantity. In fact, the specification makes clear that it is the overlap region – and thus also the interframe interval S_s – of the SOLA method that was "variable," unlike the "fixed" synthesis parameters of the purportedly inventive SOLAFS method:

> [The SOLA] method results in ***a region of overlap which is dynamic between windows*** and which requires evaluation of a cross-correlation

> with a ***variable*** number of points. As a result, ***this method allows one to change the relative overlap between windows*** which, in turn, modifies the time-scale of the input signal without significantly affecting the periods in the signal.

(*Id.* at 3:25-32 (emphasis added).)

120. Second, as described above at Section VI.B.2, a person of ordinary skill in the art would understand that, because S_S is a parameter of the purportedly inventive SOLAFS algorithm, it must be fixed in order to establish the conditions for a particular mode of time-scale modification.

E. "determined by"

Claim Nos.	Proposed Term	Apple's Proposed Construction	EPL's Proposed Construction
1, 19	determined by	"uniquely specified by"	*Plain and Ordinary Meaning*

121. A person of ordinary skill in the art reading the term "determined by" in the context of the asserted '769 claims would understand this term to mean "uniquely specified by."

1. The '769 Patent Claims, Specification, and Figures

122. In the context of the Asserted Claims, the disputed term is recited in the context of relating specific parameters to one another. Specifically, the term "determined by" relates "W_{ov}" and "the number [of signal representations] appended", on the one hand, to "W" and/or "the time-scale modification", on the other hand. In the Asserted Claims' general context (claims directed to mathematical algorithms) and specific context (defining the mathematical relationships between parameters in those algorithms), a person of ordinary skill in the art would understand that "determined by" means "uniquely specified by."

123. Notably, the Asserted Claims recite "determined by," and do not use the words "determined in part by" or "is a function of." A person of ordinary skill in the art would thus recognize that the claim language itself indicates that "determined by" means "uniquely specified by." Thus, for example, where claim 1 recites that "W_{ov} is determined by W and the time-scale modification," a person of ordinary skill in the art would understand that – for given values of W and the time-scale modification – W_{ov} would have only one unique value specified by W and the time-scale modification.

124. A person of ordinary skill in the art would further understand that "determined by" means "uniquely specified by" in view of the specification and figures. In particular, as discussed above, the '769 specification and figures explain in detail the parameters that define the conditions of the purportedly inventive SOLAFS algorithm. (*See. e.g.* '769 patent at Figs. 2, 3, 4, 8; 7:7 – 8:8.) These disclosures would confirm for a person of

ordinary skill in the art that the term "determined by" pertains to a mathematical relationship that means "uniquely specified by."

2. Extrinsic Evidence

125. The extrinsic evidence further confirms that "determined by" – particularly in the mathematical algorithm context of the '769 patent – means "uniquely specified by."

126. For example, Schwartzman, *The Words of Mathematics: An Etymological Dictionary of Mathematical Terms Used in English* (The Mathematical Assoc. of Am., 1996 ("Schwartzman") explains that the etymology of "determine" is Latin *de* ("of, from") and *terminus* (boundary)." Schwartzman defines "determine" as follows:

> Something which is determined has a definite boundary or value. In geometry, when we say that two points determine a straight line, we mean that those two points specify the line uniquely: the "boundaries" of the line have been clearly "marked."

(Schwartzman at 70.)

127. According to the Oxford English Dictionary:

> "determine . . . 11. To ascertain definitely by observation, examination, calculation, etc. (a point previously unknown or uncertain); to fix as known"

(The Oxford English Dictionary, 2nd Edition, 1989 at 550.)

128. According to the New Oxford American Dictionary:

> "**determine** . . . **2** ascertain or establish exactly, typically as a result of research or calculation . . . Mathematics specify the value, position, or form (of a mathematical or geometric object) uniquely."

(The New Oxford American Dictionary, 2001 at 466.)

129. And according to the American Heritage Dictionary:

> "**determine** . . . **2** To establish or ascertain definitely, as after consideration, investigation, or calculation. . . . **7** *Math*. To fix or define the position, form, or configuration of."

(The American Heritage Dictionary, 2nd College Edition, 1991 at 368.)

In short, each of these definitions confirms that, in a mathematical context like the '769 patent claims, "determined by" means "uniquely specified by."

3. EPL's Proposed Construction

130. EPL's proposed "plain meaning" interpretation of "determined by" does not provide any guidance as to what "determined by" should mean. As set forth above, because "determined by" means "uniquely specified by" in the context of the '769 patent and its claims, EPL's proposal is incorrect.

F. "time-scale modification" / "time-scale modifying"

Claim Nos.	Proposed Term	Apple's Proposed Construction	EPL's Proposed Construction
1, 10, 19	time-scale modification / time-scale modifying	"a change to a signal's rate of reproduction without modifying its pitch" / "changing a signal's rate of reproduction without modifying its pitch"	"speeding up or slowing down the playback rate"

131. A person of ordinary skill in the art reading the term "time-scale modification" in the context of the asserted '769 claims would understand this term to mean "a change to a signal's rate of reproduction without modifying its pitch."

132. A person of ordinary skill in the art reading the term "time-scale modifying" in the context of the asserted '769 claims would understand this term to mean "changing a signal's rate of reproduction without modifying its pitch."

1. The '769 Patent Claims, Specification, and Figures

133. The '769 patent makes clear that the "time-scale modification" recited in its claims refers to "chang[ing] a signal's rate of reproduction without modifying its pitch." For example, the specification's "TECHNICAL FIELD OF THE INVENTION" explains:

> The present invention relates to a method for time-scale modification ("TSM"), i.e., changing the rate of reproduction, of a signal and, in particular, to a method for time-scale modification of a sampled signal to provide reproduction of the signal at a wide variety of playback rates ***without an accompanying change in local periodicity***.

('769 patent at 1:6-13 (emphasis added).)

134. Likewise, the specification's "BACKGROUND OF THE INVENTION" that the "time-scale modification" of the '769 invention pertains to "chang[ing] a signal's rate of reproduction without modifying its pitch":

> A need exists in the art for a method for time-scale modification of acoustic signals such as speech or music and, in particular, a need exists for such a method which will provide time-scale modification ***without modifying the pitch or local period of the time-scale modified signals***. ***Thus, a need exists for a method for changing the perceived rate of articulation while ensuring that the local pitch period of the resulting signal remains unchanged, i.e., there are no "Alvin the Chipmunk" effects***, and that no audible splicing, reverberation, or other artifacts are introduced.

(*Id.* at 1:16-26 (emphasis added).)

135. The specification further explains that the purported '769 invention would satisfy a need in multiple contexts in which it would be important to "chang[e] a signal's rate of reproduction without modifying its pitch":

> . . . TSM can be used: (a) by a radio station to speed up dance music; (b) by a blind person to speed up a recorded lecture; (c) by a student of a foreign language to slow down instructional material; (d) by an editor to synchronize a dubbed sound track with a video signal and to compress them into convenient time slots; (e) by a secretary to slow down or speed up a dictation tape for transcription; (f) by a voicemail system to provide a message to a listener at a faster or slower rate than that at which the message was recorded; and so forth.

(*Id.* at 1:32-43; *see also id.* at 4:44-48 ("As a result, there is a need in the art for a method for modifying the time-scale of speech, music, or other acoustic material ***without modifying the pitch***, which is robust, and which does not require excessive amounts of computation.").)

136. Having identified the need in the field, the '769 patent makes clear that its purported invention is a method for "chang[ing] a signal's rate of reproduction without modifying its pitch":

> SUMMARY OF THE INVENTION
>
> Embodiments of the present invention advantageously satisfy the above-identified need in the art and provide ***a method for modifying the time-scale of speech***, music, or other acoustic material over a wide range of compression and expansion ***without modifying the pitch***.

(*Id.* at 4:44-56 (emphasis added).)

137. Likewise, the '769 patent's "DETAILED DESCRIPTION" confirms that its purported invention is a method for "chang[ing] a signal's rate of reproduction without modifying its pitch":

> DETAILED DESCRIPTION

> The present invention relates to a method for ***time-scale modification (“TSM”)***, i.e., changing the rate of reproduction, of a signal and, in particular, to a method for time-scale modification of a sampled signal by time-domain processing the sampled signal to provide reproduction of the signal at a wide variety of rates ***without an accompanying change in pitch***. An input to the inventive method is a stream of digital samples which represent samples of a signal.

(*Id.* at 6:46-55.)

2. Extrinsic Evidence

138. The extrinsic evidence further confirms that these disputed terms mean “chang[ing] a signal’s rate of reproduction without modifying its pitch.”

139. For example, the Hejna Thesis – with respect to which ’769 inventors Hejna and Musicus are the author and “Thesis Supervisor,” respectively – states:

> ***Time-scale modification*** of speech refers to processing performed on speech signals that changes the perceived rate of articulation ***without affecting the pitch*** or intelligibility of the speech.

(Hejna Thesis at 1 (emphasis added).)

140. Likewise, a published article by the ’769 inventors pertaining to the purportedly inventive SOLAFS algorithm of the ’769 patent states:

> The ***key issues*** in designing time-scale modification (TSM) systems are that ***the local pitch period remains unchanged (no Donald Duck or Minnie Mouse effects)***, and that no audible splicing, reverberation, or other artifacts are introduced.

(Hejna and Musicus, “The SOLAFS Time-Scale Modification Algorithm,” *BBN*, July 1991, at 1 (emphasis added); *see also* Roucos and Wilgus 1985 at 493 (“In ***time-scale modification***, we wish to modify the perceived rate of speech ***while preserving*** the formant structure (for intelligibility) and ***the perceived pitch (for naturalness)***.”) (emphasis added).)

141. Likewise, my own book – which was written and published before EPL filed its lawsuit against Apple – recognizes that “time-scale modification” is a term of art in the relevant field and essentially defines “time-scale modification” to mean “chang[ing] a signal’s rate of reproduction without modifying its pitch”:

> *Time Scale Modification* (TSM) means speeding up or slowing down a sound *without affecting the frequency content,* such as the perceived pitch of any tonal components. For example, TSM of speech should sound like the speaker is talking at a slower or faster pace, without distortion of the

spoken vowels. Similarly, TSM of music should change timing but not tuning.

(Smith, Julius O., *Spectral Audio Signal* Processing, W3K Publishing, 2011 at 375 (emphasis in original).)

3. EPL's Proposed Construction

142. For the reasons forth above, EPL's proposed construction is incomplete because it fails to specify that "time-scale modification" / "time-scale modifying" must be performed without modifying the pitch of the audio during playback.

VII. THE '903 PATENT FAMILY DISCLOSURE

143. The '903 Patent, '050 Patent, and '720 Patent ("the '903 Patent family") are all continuations-in-part of a patent application filed in late 2001. The '903 patent issued on March 23, 2010, the '050 patent family issued on January 1, 2013, and the '720 patent issued on February 26, 2013.

144. The '903 Patent Family relate to how time (i.e. "position") in a media file is tracked when the file is played at a variable rate of speed in digital rendering systems. For instance, when a twenty-minute audio clip is played at normal speed, the position of the audio clip after five minutes of play is five minutes. But if the audio clip is played back at twice the normal speed, the position of the audio clip after five minutes of play is ten minutes. Thus, when the playback speed is altered, the position in the audio clip at any point in time is not the same as the time elapsed since the beginning of the playback of the audio file. The '903 Patent Family claims methods for tracking this position in the audio (or other media) clip when the playback speed is varied.

145. The methods described in the '903 Patent Family refer to three different types of time. The first is "Current Time," which the '903 patent describes as conflating two different properties of media playback—"Presentation Time" and "Data Time." ('903 Patent at 1:50-2:19.)

146. "Presentation Time" is defined as the time elapsed since the beginning of the media content presentation: "The first property of media playback that is conflated in the concept of Current Time is a time elapsed since the beginning of the media content presentation (hereafter called "Presentation Time"). Thus, if the media has been playing for one minute, the value of Presentation Time is 60,000 milliseconds." ('903 Patent at 1:50-2:19.) When the playback speed of media is varied, calculations of the current position in the media content in many media players, which regularly rely on "Presentation Time," will not be accurate.

147. The '903 Patent Family attempt to solve this problem by using, in addition to Presentation Time, what it calls "Data Time." The patents define Data Time as an amount of time required to render the portion of the temporal sequence presentation data at a normal playback rate:

> In a traditional linear media stream that is always played back at a fixed, "normal" rate, any given content element is always presented after a fixed amount of time has elapsed from the beginning of playback. Because of this, each such content element can be regarded as having a timestamp associated with it, i.e., a time value specifying how long it would take to reach that location, starting from the beginning of the media content, and playing at normal rate. Hereinafter we will call this property "Data Time."

('903 Patent at 1:66-2:7.) When the rate of playback of a media file is varied, Presentation Time and Data Time are not the same.

148. The '903 Patent family discloses a variety of methods, devices and systems relating to rendering presentation data. Claim 12 of the '903 Patent is illustrative of the asserted claims across all three patents in the '903 Patent Family (the disputed claim term is indicated in **bold**):

> 12. A method, performed by at least one machine, for rendering temporal sequence presentation data in a machine-implemented rendering system, the temporal sequence presentation data being tangibly stored in a first computer-readable medium, the method comprising steps of:
>
> (A) receiving a request from a first component of the rendering system for a first **current time**;
>
> (B) maintaining a value of a presentation time parameter tangibly stored in a second computer-readable medium and representing an amount of time elapsed during rendering of a portion of the temporal sequence presentation data by the rendering system;
>
> (C) providing the value of the presentation time parameter to the first component in response to the request;
>
> (D) receiving a request from a second component of the rendering system for a second **current time**;
>
> (E) maintaining a value of a data time parameter tangibly stored in a third computer-readable medium and representing an amount of time required to render the portion of the temporal sequence presentation data at a default presentation rate; and
>
> (F) providing the value of the data time parameter to the second component in response to the request;
>
> wherein the value of the presentation time parameter is not equal to the value of the data time parameter.

('903 Patent, claim 12.)

VIII. DISPUTED CLAIM TERMS OF THE '903 PATENT FAMILY

A. "current time"

Claim Nos.	Proposed Term	Apple's Proposed Construction	EPL's Proposed Construction
'903 patent: claims 4, 12, 22 '050 patent: claims 4, 16 '720 patent: claims 1, 6, 7, 11	current time	"measure of time that is unresolved as to whether the rendering system should return a presentation time parameter value or a data time parameter value"	"a current position in the media content that can be expressed either as the time elapsed since the beginning of the media content presentation or as a location in the media content stream that is currently being played"

149. A person of ordinary skill in the art reading the term "current time" in the context of the asserted '903 Patent Family claims would understand this term to mean "measure of time that is unresolved as to whether the rendering system should return a presentation time parameter value or a data time parameter value." This meaning is compelled by the claim language and the clear teaching of the specifications.

1. Claim Language

150. Each of the asserted claims in the '903 Patent Family that includes the term "current time" does so in the context of a request from a component of the rendering system for the "current time." Indeed, every occurrence of "current time" in the asserted claims appears as part of the phrase "*request* . . . for a current time." ('903 Patent, claims 4, 12, 22; '050 patent, claims 4, 16; '720 patent, claims 1, 6, 7, 11.) And in every one of these asserted claims, in response to the "request . . . for a current time," the rendering system returns one of a presentation time parameter value or a data time parameter value. (*Id.*)

151. The claim language in each of the asserted claims in which "current time" appears makes clear that "current time" is not itself a time value that the rendering system maintains or calculates; rather the claim language is clear that "current time" defines the particular request from the rendering system as one in which the rendering system seeks a measure of time that is one of the presentation time parameter value or the data time parameter value. (*Id.*)

152. The claim language also makes clear that the "request . . . for current time," when made, is unresolved as to whether the rendering system need return the presentation time parameter value or the data time parameter value. Were the "request . . . for current time" not unresolved in this respect, a "request . . . for current time" would make little sense; in that instance, the claims could simply recite a "request . . . for *presentation*

time" or "request . . . for *data time*." That the claims do not use this phrasing, and instead recite "request . . . for *current time*," supports Apple's proposed construction.

153. In addition, two of the asserted claims in which "current time" appears include a limitation requiring that the rendering system "determin[e], based on a property of the request [for current time], whether to return a value of a data time parameter or a value of a presentation time parameter." ('720 Patent, claims 7, 11.) Based on this language, "current time" must be unresolved as to whether the rendering system should return a presentation time parameter value or a data time parameter value.

2. Intrinsic Evidence

154. The '903 Patent Family specifications also support Apple's proposed construction of "current time."

155. The specifications describe that the concept of "current time" is not straightforward when tracking time in the context of media players that contain TSM functionality. ('903 Patent at 1:50-2:35.) This is because "current time" "conflates two subtly different properties of media playback." ('903 Patent at 1:50-52.) The first of these properties is presentation time, which the specification describes as "a time elapsed since the beginning of the media content presentations). ('903 Patent at 1:52-62.) The second of the properties is data time, which the specification describes as "a time value specifying how long it would take to reach [the present] location [of the media content], starting from the beginning of the media content, and playing at a normal rate." ('903 Patent at 1:63-2:7.)

156. In the context of the alleged inventions of the '903 Patent Family—media players with TSM functionality—presentation time and data time are not necessarily equal. ('903 Patent at 1:50-2:35.) Thus, "current time," which conflates these two unequal time values, is not itself a calculated amount of time. Rather, it is a measure of time that is one of a presentation time parameter or a data time parameter. (*Id.*; *see also* 3:1-3, 4:19-61, 16:34-49, 16:62-17:1.)

157. Furthermore, the '903 Patent Family specifications explicitly describe that one of the benefits of the alleged invention of the '903 Patent is improved performance as a result of the rendering system making a determination regarding which value, "presentation time" or "data time," to return in response to a request for "current time":

> In summary, in various embodiments of the present invention in which a traditional player system is enhanced to support variable speed playback, the ***performance of the system may be improved by ensuring that the Timing Renderer determines whether to base the time value returned on the Current Presentation Time or the Current Data Time*** when a component asks for the Current Time.

('903 Patent, col. 16:62-17:1 (emphasis added).) For the rendering system to make this determination, the request for "Current Time" must be *unresolved* as to whether to base the time value returned on the Current Presentation Time or the Current Presentation

Time. If that were not the case, there would be no determination for the rendering system to make.

3. EPL's Proposed Construction

158. EPL's proposed construction is inconsistent with both the plain language of the claims and the specifications of the '903 Patent Family. EPL's proposed construction allows for "current time" to be "expressed *either* as the time elapsed since the beginning of the media content presentation or as a location in the media content stream that is currently being played." As described in detail above, however, the claim language and specifications are clear that in response to the "request . . . for current time," the rendering system returns *one of* the presentation time parameter value or the data time parameter value, not a time value that can be expressed as *either* the presentation time parameter value or the data time parameter value. ('903 Patent, claims 4, 12, 22; '050 patent, claims 4, 16; '720 patent, claims 1, 6, 7, 11.) EPL's proposed construction would create an inconsistency in the claims between the "request . . . for current time"—according to EPL, a request for a current position in the media content that can be expressed as *either* of two different time values—and the response to that request—unambiguously *one of* the presentation time parameter value or the data time parameter value.

159. In addition, EPL's proposed construction is inconsistent with the aim of the invention of the '903 Patent Family. The '903 Patent Family specifications explain that an ambiguous time value is one of the primary problems that the '903 Patent Family was aimed at solving. ('903 Patent, 2:20-30.) But EPL's proposed construction creates this same ambiguous time value—one that can be expressed as *either* of two different time values—that the inventors identified as a problem in the prior art.

160. EPL's construction of "current time" is also deficient in that the phrase "a current position in the media content" is ambiguous. Although the specifications state that "Current Time is, in effect, a current 'position' in the media content that is being displayed and rendered," ('903 Patent at 1:27-29), that language is not a precise definition of "current time." In particular, the cited language from the specifications includes the words "in effect." Moreover, the word "position" is in quotation marks. At best, this phrase that EPL includes in its construction is a loose approximation of what "current time" represents. The claim language and the specifications, however, make clear that a "request . . . for current time" is a request for a measure of time.

161. Similarly, EPL's construction is ambiguous in its use of the phrase "a location in the media content stream that is currently being played." The language of the relevant claims make clear that in response to a "request . . . for current time," the rendering system returns one of the presentation time parameter value or the data time parameter value. It is my understanding that the parties have agreed that "data time parameter" means "an amount of time required to render the portion of the temporal sequence presentation data at a normal playback rate." In my opinion, the proper construction of "current time" should include either "data time" or the agreed upon definition. The phrase "a location in the media content stream that is currently being played," while

taken from the specifications, is imprecise, makes the construction confusing, and is inconsistent with the language of the relevant claims.

IX. HEARING AND TRIAL EXHIBITS

162. I may rely on visual aids and demonstrative exhibits that demonstrate the bases of my opinions. Examples of these visual aids and demonstrative exhibits may include, for example, claim charts, patent drawings, excerpts from patent specifications, file histories, interrogatory responses, deposition testimony and deposition exhibits, as well as charts, diagrams, videos and animated or computer-generated video.

163. Other than as referred to in this report, I have not yet prepared any exhibits for use at trial as a summary or support for the opinions expressed in this report, but I expect to do so in accordance with the Court's scheduling order.

X. SUPPLEMENTATION OF OPINIONS

164. I reserve the right to supplement the opinions I have offered in this declaration in light of any critique or alternative declarations proffered on behalf of EPL.

Dated: September 21, 2013

Julius O. Smith III

Prof. Julius O. Smith, Ph.D.

Exhibit A

Curriculum Vitae

JULIUS O. SMITH III

Center for Computer Research in Music and Acoustics (CCRMA)
Department of Music, Stanford University, Stanford, California 94305 USA

July 2013

Education

Ph.D., Electrical Engineering, Spring 1983, Stanford University.
M.S./E.E., Statistical Signal Processing, Spring 1978, Stanford University.
B.S./E.E., Control Communications and Circuits, Spring 1975, Rice University.

Work Experience

Fall 2004 to the present: Professor of Music and Associate Professor (by courtesy) of Electrical Engineering, Stanford University, based at the Center for Computer Research in Music and Acoustics (CCRMA). Activities include teaching courses in signal processing and music technology, graduate student advising, and research in signal processing techniques applied to music and audio.

Fall 1994 to Fall 2004: Associate Professor of Music and (by courtesy) Electrical Engineering, Stanford University, at CCRMA. Activities as listed above.

Fall 1989 to Summer 1994 (part to full time): Associate Professor (Research) at CCRMA. Activities included basic research in signal processing techniques applied to music and audio, teaching signal processing courses in support of my research program (Music 320/420/421, EE 265/266), and supervising Ph.D. research in the Computer Music Ph.D. program at CCRMA and in the Electrical Engineering Department.

Summer 1986 to Winter 1993 (part to near full time): Software engineer at NeXT Inc., responsible for signal processing software pertaining to music and audio: Designed and implemented a variable-rate transform coder for real-time compression/decompression of "CD-quality" audio signals on the Motorola DSP56001 signal processing chip. Managed the Sound, Music, and Signal Processing Group at NeXT from its inception in 1986 until 1992. Managed four outside consultants on software development projects. Co-designed the object-oriented NeXT Music Kit with David Jaffe. Designed and implemented DSP56001 software supporting the NeXT Music Kit, including the real-time DSP monitor and unit-generator modules for sound synthesis and signal processing. Wrote and supported The NeXT DSP Library. Helped support and debug the Sound/DSP Mach driver and the NeXT Sound Library.

Fall 1984 to Summer 1986 (half time): Research Associate, CCRMA, Stanford. Projects included violin modeling, woodwind modeling, new digital filter design methods tailored to audio applications, new reverberation techniques, time-varying sampling-rate conversion, digital filtering

software, spectrum analysis software, system identification software, and pitch detection. Duties included teaching a two-year sequence in digital signal processing aimed at graduate students interested in music applications of signal processing and acoustics.

Fall 1982 to Summer 1986 (half time): Systems Control Technology, Palo Alto CA. Projects included time-delay estimation, ARMA modeling and spectrum analysis, underwater acoustic signal processing, HF communications signal processing, and general tool development.

Honors (reverse chronological order)

- CIRMMT Distinguished Lecture, McGill University, Sep. 2010
- Keynote Speaker, Digital Audio Effects Conference (DAFx-2009, Como Italy)
- Fellow, Audio Engineering Society, 2008
- Heyser Lecture, Audio Engineering Society Conference (AES-2006, San Francisco)
- Invited Masterclass, Audio Engineering Society Conference (AES-2006, San Francisco)
- Keynote Speaker, Digital Audio Effects Conference (DAFx-2006, Montreal, Canada)
- Keynote Speaker, IEEE Workshop on Applications of Signal Processing to Audio & Acoustics (WASPAA-2005)
- Invited presentation: American Association of Physics Teachers, Sacramento, Aug. 2004
- Fellow of the Acoustical Society of America, 2003:
  "For applications of digital signal processing to musical acoustics"
- Invited Speaker, first in the Opening Session, Stockholm Musical Acoustics Conference, 2003
- Invited Tutorial on Virtual Musical Instrument Design, Acoustical Society of America, 2000
- Technical Program Chair, IEEE Audio & Acoustics Signal Processing Workshop, 1997
- IRCAM Scientific Council, since 1996
- Plenary Speaker, Nordic Acoustics Conference, 1996
- Keynote Speaker, Tempo Reale Workshop on Physical Modeling 1996
- Inventor Recognition Award, Stanford Office of Technology and Licensing, 1996
- Keynote Speaker, ICMC-91 (Int. Computer Music Conf.), Montreal
- Invited Speaker, Acoustical Society of America (several meetings)
- Sigma Xi, 1983
- Hertz Graduate Fellowship, Fall 1977 to Fall 1982
- Magna Cum Laude, Rice University, 1975
- Tau Beta Pi, 1974
- Brown Engineering Merit Award, 1974 and 1973
- American Legion Award, 1971
- Moody Foundation Scholarship, 1971

Patents

Below are issued U.S. patents, assigned to Stanford University, on which I am a named inventor. There are a few additional patents associated with my outside consulting activities. Foreign patents are not listed.

- U.S. Patent No. 4,984,276: Artificial reverberation using digital waveguide networks.
- U.S. Patent No. 5,029,509: Spectral modeling synthesis.

- U.S. Patent No. 5,212,334 & 5,448,010: Digital sound synthesis based on acoustic models.
- U.S. Patent No. 5,331,587: Clipped signal restoration.
- U.S. Patent No. 5,353,372: Tracking pitch detector.
- U.S. Patent No. 5,466,884: Wave-digital piano hammer.
- U.S. Patent No. 5,471,007 and 5,614,686: Multidimensional waveguide synthesis.
- U.S. Patent No. 5,471,686: 2D digital waveguide synthesis.
- U.S. Patent No. 5,500,486: Commuted digital waveguide synthesis.
- U.S. Patent No. 5,587,548: CIP for commuted waveguide synthesis.
- U.S. Patent No. 5,777,255: Commuted piano synthesis.
- U.S. Patent No. 5,701,393: One-multiply digital-waveguide sinusoidal oscillator.
- U.S. Patent No. 5,781,461: Legato waveguide synthesis.
- U.S. Patent No. 6,284,965: Truncated IIR horn modeling.
  6284965

Expert Witness and Patent Analysis Engagements

Parties: Personal Audio v. Samsung
Time frame: 2012
Primary contact: Evan S. Day
Contact's firm: Perkins Coie
Contribution: Non-testifying consulting work regarding noninfringement of music player produc
Relevant expertise: Knowledge of music playlist technology and software

Parties: LSI v. Barnes & Noble
Time frame: 2012
Primary contact: Charles Basinger
Contact's firm: Quinn Emanuel Urquhart & Sullivan, LLP
Contribution: Expert-witness work regarding computer audio buffers and FIFOs
Relevant expertise: Digital audio hardware and software prior art

Parties: MobileMedia Ideas v. HTC
Time frame: 2011
Primary contact: Cary Chien
Contact's firm: McDermott Will & Emery LLP
Contribution: Non-testifying patent claim analysis and tutorial preparation
Relevant expertise: Knowledge of digital audio compression technologies and MPEG audio systems in particular.

Parties: [Confidential]
Time frame: 2009–2010
Primary contact: [Confidential]
Contact's firm: Paul Hastings, LLP
Contribution: Non-testifying consulting work and exhibit/demo/tutorial preparation
Relevant expertise: Knowledge of music-related technologies and software engineering in general.

Parties: Patent portfolio analysis
Time frame: 2008
Primary contact: Laura Majerus
Contact's firm: Google Inc.
Contribution: Due-diligence consulting regarding audio CODEC systems.
Relevant expertise: Knowledge of digital audio signal processing technologies used in Internet audio CODECs.

Parties: MOAEC, Inc. v. Napster, L.L.C.
Time frame: 2008–2009
Primary contact: John Cotter
Contact's firm: K & L Gates, LLP
Contribution: Testifying witness regarding systems for organizing and playing music
Relevant expertise: Knowledge of technologies for music storage and playback.

Parties: Friskit, Inc. v. RealNetworks, Inc. and Listen.com
Time frame: 2004–2008
Primary contact: Charles K. Verhoeven
Contact's firm: Quinn Emanuel Urquhart Oliver & Hedges, LLP
Contribution: Testifying witness regarding systems for streaming audio over the Internet
Relevant expertise: General familiarity with audio signal processing and software technologies.

Parties: Apple Computer v. Burst.com
Time frame: 2007
Primary contact: Nicholas V. Martini
Contact's firm: Weil, Gotshal & Manges LLP
Contribution: Assisted with obtaining prior art.
Relevant expertise: Managed the Sound and Music Group at NeXT Computer from its inception in mid-1986.

Parties: Panasonic/Matsuhita v. Media Tek, Inc.
Time frame: 2006
Primary contact: Arlyn Alonzo
Contact's firm: McDermott Will & Emery LLP
Contribution: Expert witness regarding vibration-immune audio CD playback systems
Relevant expertise: General familiarity with digital audio signal processing and associated format conversions, error correction, and the like.

Parties: SSI International v. a former contractor
Time frame: 2002–2004
Primary contact: R.W. Tate
Contact's firm: Robert Tate Atty at Law
Contribution: Technical analysis and support on behalf of SSI
Relevant expertise: Time-scale modification and associated technology, software engineering practices

Parties: Play-Media v. AOL Time Warner
Time frame: Autumn 2001
Primary contact: Michael T. Zeller
Contact's firm: Quinn Emanuel et al., LLP
Contribution: Declaration, federal court testimony on behalf of AOL
Relevant expertise: Software engineering and multimedia technology, audio compression

Parties: Creative Labs v. Aureal Semiconductor
Time frame: Latter half of 1999
Primary contact: Peter Detre
Contact's firm: Munger Tolles & Olson LLP
Contribution: Non-testifying contributor to declarations of attorneys and other experts
Relevant expertise: 3D sound technology

Parties: (confidential arbitration)
Time frame: Summer 1998
Primary contact: (available upon request)
Contact's firm:
Contribution: Declaration pertaining to competing patent portfolios
Relevant expertise: 3D sound technology

Parties: Polycom v. U.S. Robotics
Time frame: Spring 1996
Primary contact: Eric Wesenberg
Contact's firm: Brobeck, Pheger & Harrison LLP (now at Orrick, Herrington et al. LLP)
Contribution: Scientific analysis, declaration, and deposition
Relevant expertise: Acoustic functionality of conference telephones

Parties: Yamaha v. ESS
Time frame: Earlier
Primary contact: David Fehrman
Contact's firm: Graham & James LLP
Contribution: Patent claims analysis and expert witness declaration
Relevant expertise: FM sound synthesis technology

Publications

Below are my publications in approximate chronological order.

Hyperlinks to PDFs, when available, are typically attached only to the `http:` portion of each URL (to avoid line-wrapping problems in LaTeX).

[1] J. O. Smith and J. B. Allen, "Variable bandwidth adaptive delta modulation", *Bell System Technical Journal*, vol. 60, no. 5, pp. 719–737, May-June 1981.

[2] J. O. Smith and J. B. Angell, "A constant-gain digital resonator tuned by a single coefficient", *Computer Music Journal*, vol. 6, no. 4, pp. 36–40, 1982.

[3] M. Gutknecht, J. O. Smith, and L. N. Trefethen, "The Caratheodory-Fejer (CF) method for recursive digital filter design", *IEEE Transactions on Acoustics, Speech, Signal Processing*, vol. 31, no. 6, pp. 1417–1426, 1983.

[4] J. O. Smith, "Synthesis of bowed strings", in *Proceedings of the 1982 International Computer Music Conference, Venice*. 1982, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, Essentially fully contained in [5].

[5] J. O. Smith, *Techniques for Digital Filter Design and System Identification with Application to the Violin*, PhD thesis, Elec. Engineering Dept., Stanford University (CCRMA), June 1983, CCRMA Technical Report STAN-M-14, `http://ccrma.stanford.edu/STANM/stanms/stanm14/`.

[6] J. O. Smith, "Spectral pre-processing for audio digital filter design", in *Proceedings of the 1983 International Computer Music Conference, Eastman School of Music*. 1983, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, Essentially fully contained in [5].

[7] J. O. Smith, "An allpass approach to digital phasing and flanging", in *Proceedings of the 1984 International Computer Music Conference, Paris*. 1984, pp. 103–109, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, CCRMA Technical Report STAN-M-21, `http://ccrma.stanford.edu/STANM/stanms/stanm14/`.

[8] J. O. Smith and P. Gossett, "A flexible sampling-rate conversion method", in *Proc. 1984 Int. Conf. Acoustics, Speech, and Signal Processing (ICASSP-84), San Diego*, New York, Mar. 1984, vol. 2, pp. 19.4.1–19.4.2, IEEE Press, expanded tutorial and associated free software available at the Digital Audio Resampling Home Page: `http://ccrma.stanford.edu/~jos/resample/`.

[9] J. O. Smith and B. Friedlander, "Estimation of multipath delay", in *Proc. 1984 Int. Conf. Acoustics, Speech, and Signal Processing (ICASSP-84), San Diego*, New York, Mar. 19–21 1984, pp. 15.9.1–15.9.4, IEEE Press.

[10] J. O. Smith and B. Friedlander, "Global convergence of the constant modulus algorithm", *Proceedings of the International Conference on Acoustics, Speech, and Signal Processing, Tampa, Florida*, pp. 30.5.1–30.5.4, 1985.

[11] J. O. Smith and B. Friedlander, "Extensions of the constant modulus algorithm", *Asilomar-84*, 1984.

[12] J. O. Smith and B. Friedlander, "Adaptive multipath delay estimation", *IEEE Transactions on Acoustics, Speech, Signal Processing*, vol. 33, no. 4, pp. 812–822, Aug. 1985.

[13] J. O. Smith and B. Friedlander, "Adaptive interpolated time-delay estimation", *IEEE Transactions on Aerospace and Electronic Systems*, vol. 21, no. 2, pp. 180–199, Mar. 1985.

[14] J. O. Smith, "A new approach to digital reverberation using closed waveguide networks", in *Proceedings of the 1985 International Computer Music Conference, Vancouver*. 1985, pp. 47–53, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, also available in [19].

[15] C. Chafe, D. Jaffe, K. Kashima, B. Mont-Reynaud, and J. O. Smith, "Techniques for note identification in polyphonic music", in *Proceedings of the 1985 International Computer Music Conference, Vancouver*. 1985, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`.

[16] J. O. Smith, "Elimination of limit cycles and overflow oscillations in time-varying lattice and ladder digital filters", in *Proceedings of the IEEE Conference on Circuits and Systems, San Jose*, May 1986, pp. 197–299, conference version; full version available in [19].

[17] J. O. Smith, "Efficient simulation of the reed-bore and bow-string mechanisms", in *Proceedings of the 1986 International Computer Music Conference, The Hague*. 1986, pp. 275–280, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, also available in [19].

[18] J. O. Smith, "Waveguide filter tutorial", in *Proceedings of the 1987 International Computer Music Conference, Champaign-Urbana*. 1987, pp. 9–16, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`.

[19] J. O. Smith, "Music applications of digital waveguides", Tech. Rep. STAN-M-39, CCRMA, Music Department, Stanford University, 1987, CCRMA Technical Report STAN-M-39, `http://ccrma.stanford.edu/STANM/stanm39/`.

[20] J. O. Smith and J. S. Abel, "Closed-form least-squares location estimation from range-difference measurements", *IEEE Transactions on Acoustics, Speech, Signal Processing*, vol. 35, no. 12, pp. 1661–1669, Dec. 1987.

[21] J. O. Smith and X. Serra, "PARSHL: A program for the analysis/synthesis of inharmonic sounds based on a sinusoidal representation", in *Proceedings of the 1987 International Computer Music Conference, Champaign-Urbana*. 1987, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, Also available as Stanford Music Department Technical Report STAN-M-43. Expanded version available on-line at `http://ccrma.stanford.edu/~jos/parshl/`.

[22] J. O. Smith, D. A. Jaffe, and L. Boynton, "Music system architecture on the NeXT computer", *The Proceedings of the AES 7th International Conference: Audio in Digital Times*, pp. 301–312, May 14–17 1989.

[23] J. O. Smith, "Computer music on the DSP56001", in *Proceedings of the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, NY*, New York, Oct. 1989, IEEE Press.

[24] C. Roads, Ed., *The Music Machine*, MIT Press, Cambridge, MA, 1989.

[25] J. O. Smith, "Unit-generator implementation on the NeXT DSP chip", in *Proceedings of the 1989 International Computer Music Conference, Ohio*. 1989, pp. 303–306, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`.

[26] J. O. Smith, "Rectangular, Hanning, and Hamming window transforms", Winter 1992, Mathematica notebook for Music 420 (EE 367A), `URL:ftp://ccrma-ftp.stanford.edu/pub/DSP/Tutorials/GenHamming.ma.Z`.

[27] J. O. Smith, "The Kaiser window", Winter 1992, Mathematica notebook for Music 420 (EE 367A), URL:`ftp://ccrma-ftp.stanford.edu/pub/DSP/Tutorials/Kaiser.ma.Z`.

[28] J. O. Smith, "The window method for digital filter design", Winter 1992, Mathematica notebook for Music 420 (EE 367A), URL:`ftp://ccrma-ftp.stanford.edu/pub/DSP/Tutorials/Kaiser.ma.Z`.

[29] J. O. Smith, "Efficient yet accurate models for strings and air columns using sparse lumping of distributed losses and dispersion", in *Proceedings of the Colloquium on Physical Modeling*, Grenoble, France, 1990, ACROE, Essentially superceded by [34].

[30] J. O. Smith, "Viewpoints on the history of digital synthesis", in *Proceedings of the 1991 International Computer Music Conference, Montréal*. 1991, pp. 1–10, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, `http://ccrma.stanford.edu/~jos/kna/`.

[31] J. S. Abel and J. O. Smith, "Restoring a clipped signal", in *Proceedings of the International Conference on Acoustics, Speech, and Signal Processing, Toronto*, New York, May 1991, IEEE Press.

[32] J. O. Smith, "Waveguide simulation of non-cylindrical acoustic tubes", in *Proceedings of the 1991 International Computer Music Conference, Montréal*. 1991, pp. 304–307, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`.

[33] J. O. Smith and P. R. Cook, "The second-order digital waveguide oscillator", in *Proceedings of the 1992 International Computer Music Conference, San Jose*. 1992, pp. 150–153, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, `http://ccrma.stanford.edu/~jos/wgo/`.

[34] J. O. Smith, "Physical modeling using digital waveguides", *Computer Music Journal*, vol. 16, no. 4, pp. 74–91, 1992, special issue: Physical Modeling of Musical Instruments, Part I. `http://ccrma.stanford.edu/~jos/pmudw/`.

[35] J. O. Smith, "Efficient synthesis of stringed musical instruments", in *Proceedings of the 1993 International Computer Music Conference, Tokyo*. 1993, pp. 64–71, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, incorporated into [249].

[36] J. O. Smith, "Use of commutativity in simplifying acoustic simulations", in *Proceedings of the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, NY*, New York, Oct. 1993, IEEE Press.

[37] J. O. Smith and D. Rocchesso, "Connections between feedback delay networks and waveguide networks for digital reverberation", in *Proceedings of the 1994 International Computer Music Conference, Århus*. 1995, pp. 376–377, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`.

[38] J. O. Smith and S. A. Van Duyne, "Commuted piano synthesis", in *Proceedings of the 1995 International Computer Music Conference, Banff*. 1995, pp. 319–326, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, `http://ccrma.stanford.edu/~jos/pdf/svd95.pdf`.

[39] D. P. Berners and J. O. Smith, "On the use of Schrodinger's equation in the analytic determination of horn reflectance", in *Proceedings of the 1994 International Computer Music Conference, Århus*. 1994, pp. 419–422, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`.

[40] D. P. Berners and J. O. Smith, "Super-spherical wave simulation in flaring horns", in *Proceedings of the 1995 International Computer Music Conference, Banff*. 1995, pp. 112–113, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`.

[41] B. Friedlander and J. O. Smith, "Analysis and performance evaluation of an adaptive notch filter", *IEEE Transactions on Information Theory*, vol. 30, no. 2, pp. 283–295, Mar. 1984.

[42] J. W. Gordon and J. O. Smith, "A sine generation algorithm for VLSI applications", in *Proceedings of the 1985 International Computer Music Conference, Vancouver*. 1985, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, `http://ccrma.stanford.edu/~jos/pdf/GordonAndSmith86.pdf`.

[43] S. Hirschman, P. R. Cook, and J. O. Smith, "Digital waveguide modelling of reed woodwinds: An interactive development environment on the NeXT computer", in *Proceedings of the 1991 International Computer Music Conference, Montréal*. 1991, pp. 300–303, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/, available in "CCRMA Papers on Physical Modeling from the 1991 International Computer Music Conference," Department of Music Technical Report STAN-M-73, Stanford University, October 1991.

[44] D. A. Jaffe and J. O. Smith, "Extensions of the Karplus-Strong plucked string algorithm", *Computer Music Journal*, vol. 7, no. 2, pp. 56–69, 1983, Reprinted in [24, pp. 481–494].

[45] D. A. Jaffe and J. O. Smith, "Real time sound processing and synthesis on multiple DSPs using the Music Kit and the Ariel QuintProcessor", in *Proceedings of the 1993 International Computer Music Conference, Tokyo*. 1993, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/.

[46] D. A. Jaffe and J. O. Smith, "Performance expression in commuted waveguide synthesis of bowed strings", in *Proceedings of the 1995 International Computer Music Conference, Banff*. 1995, pp. 343–346, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/.

[47] D. A. Jaffe, J. O. Smith, and N. Porcaro, "The Music Kit on a PC", in *Proceedings of the First Brazilian Symposium on Computer Music, XIV Congress of the Brazillian Society of Computation, Caxambu*, Canela, Brazil, Aug. 1994, pp. 63–69, Informática UFRGS.

[48] N. Porcaro, P. Scandalis, J. O. Smith, D. A. Jaffe, and T. Stilson, "SynthBuilder—a graphical real-time synthesis, processing and performance system", in *Proceedings of the 1995 International Computer Music Conference, Banff*. 1995, pp. 61–62, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/.

[49] D. Rocchesso and J. O. Smith, "Circulant feedback delay networks for sound synthesis and processing", in *Proceedings of the 1994 International Computer Music Conference, Århus*. 1995, pp. 378–381, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/.

[50] X. Serra and J. O. Smith, "Spectral modeling synthesis: A sound analysis/synthesis system based on a deterministic plus stochastic decomposition", *Computer Music Journal*, vol. 14, no. 4, pp. 12–24, 1990, Software and recent papers URL: //www.iua.upf.es/~xserra/.

[51] X. Serra and J. O. Smith, "Sound-sheet examples for a sound analysis/synthesis system based on a deterministic plus stochastic decomposition", *Computer Music Journal*, vol. 15, no. 1, pp. 86–87, Spring 1991.

[52] S. A. Van Duyne and J. O. Smith, "Implementation of a variable pick-up point on a waveguide string model with FM/AM applications", in *Proceedings of the 1992 International Computer Music Conference, San Jose*. 1992, pp. 154–157, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/.

[53] S. A. Van Duyne and J. O. Smith, "Physical modeling with the 2-D digital waveguide mesh", in *Proceedings of the 1993 International Computer Music Conference, Tokyo*. 1993, pp. 40–47, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/, http://ccrma.stanford.edu/~jos/pdf/mesh.pdf.

[54] S. A. Van Duyne and J. O. Smith, "The 2-D digital waveguide mesh", in *Proceedings of the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, NY*, New York, Oct. 1993, IEEE Press.

[55] S. A. Van Duyne and J. O. Smith, "A simplified approach to modeling dispersion caused by stiffness in strings and plates", in *Proceedings of the 1994 International Computer Music Conference, Århus*. 1994, pp. 407–410, Computer Music Association, searchable at http://quod.lib.umich.edu/i/icmc/.

[56] S. A. Van Duyne, J. R. Pierce, and J. O. Smith, "Traveling-wave implementation of a lossless mode-coupling filter and the wave digital hammer", in *Proceedings of the 1994 International Computer Music Conference, Århus*. 1994, pp. 411–418, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, Also presented at the conference of the Acoustical Society of America, Nov., 1994.

[57] S. A. Van Duyne and J. O. Smith, "Developments for the commuted piano", in *Proceedings of the 1995 International Computer Music Conference, Banff*. 1995, pp. 335–343, Computer Music Association, searchable at `http://quod.lib.umich.edu/i/icmc/`, `http://ccrma.stanford.edu/~jos/pdf/vds95.pdf`.

[58] J. O. Smith, "Discrete-time modeling of acoustic systems", *CCRMA Associates Conference*, May 1995, Monograph in progress.

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